Hyperprism™

Real Time Sound Designing Software

Manual

Arboretum
SYSTEMS

P. O. Box 470580, San Francisco, California 94147 U. S. A.
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1

Introduction to Hyperprism™

The universe of possible musical forms is as unlimited as the number of external forms of crystals.
Edgard Varèse, speaking about his composition *Hyperprism* (1923)

This chapter introduces Hyperprism – a highly flexible *real-time* effects processor program for Apple Macintosh computers. Conceived for musical sound transformation, Hyperprism combines the ease of the Macintosh interface with the signal processing power of Digidesign™ sound cards. Its primary features are the following:

- Interactive graphical control of up to two effect parameters in real time.
- All your control movements are recorded and can be played back by Hyperprism in synchronization with the original sound file.

- Two simultaneous effects (one fixed delay and one time varying effect).

- Multipass processing means that you can apply any number of effects to a single sound file.

- Nondestructive processing means that your original sound files are always safe.

- Thru mode lets you process sounds "live," with no limits on the duration of the sounds.

For instructions on installing and configuring Hyperprism, please consult **Appendix A**.

Be sure to read the **Read Me file** on your software diskette for important additional information we were not able to include in the manual.

Hyperprism 1.1 implements no less than 21 musical effects:

1. Low pass filter
2. High pass filter
3. Band pass filter
4. Band reject filter
5. Phaser
6. Flanger
7. Chorus
8. Single delay
9. Multi delay
10. Echo
11. Ring modulator
12. Ring shifter
13. Frequency shifter
14. Pitch follower
15. Envelope follower
16. Doppler
17. Pitch Time changer
18. Tremolo
19. Vibrato
20. Stereo dynamics
21. Balance

Within each of these effects is a wide range of behavior that you can control interactively.

File Mode and Thru Mode

Hyperprism works in two modes:

File mode

In File mode, the processing applies to samples already stored on disk. These files must be in one of two formats: Sound Designer II (SDII) or Audio Interchange File Format (AIFF).

This processing is nondestructive – the original samples in the sound file are never altered. Hyperprism records only the changes in the effects parameters. It is thus possible to create as many different processing versions (called *variations*) as you want, without altering the original sound.
Having applied one or many variations to a sound file, you make the processing permanent by generating a new sound file from one of the variations. This processed file can itself be subjected to new variations. In this way you can apply many different effects to the same sound.

Thru (or direct) mode

In Thru mode, effects processing occurs as the sound samples are taken in from an external source. The incoming samples are processed and immediately passed to the output in real time (as with standard effects boxes). There is no limitation on the duration of the sound to be processed because the output is not written to a disk file.

Roadmap of this Manual

Chapter 2 is a brief overview of sound file basics.

Chapter 3 takes you on a guided tour of Hyperprism's features.

Chapter 4 serves as a reference section, describing the specifics of each effect.

Appendix A tells you how to install and configure Hyperprism.

Appendix B describes the processes of analyzing a sound's pitch and envelope with Hyperprism.

Appendix C is a guide to troubleshooting, advising you what to do when something seems to go wrong.

At the end of the manual is an **Index** to help you with particular topics.

2

Sound File Basics

This chapter introduces the basic concepts of storage requirements and file formats used by Hyperprism.

Storing Digital Audio Samples

If you sample sounds at the normal compact disc rate (44.1 KHz) in stereo, you must have the following hard disk capacity for each second of sound:

The CD sampling rate is 44100 samples per second.
Stereo means 44100 x 2.
Each 16-bit sample requires two 8-bit bytes.
Thus 44100 x 2 x 2 = 176,400 bytes or about 172 Kbytes (1 Kbyte = 1024 bytes).

For each minute of sound, this number is multiplied by 60, resulting in about 10.2 Mbytes of storage. A good rule of thumb is:

Each minute of stereo sound at 44.1 KHz requires about 10 Mbytes of disk space.

Thus, with an empty 200 Mbyte hard disk you can record a little less than 20 minutes of CD-quality stereo sound (or more precisely, 19 min, 20 sec).

Hyperprism supports the following sound file formats: Sound Designer II (SD II) and Audio Interchange File Format (AIFF), both monaural and stereophonic.

• ***Sound Designer II format (SD II)*** *– the most flexible format to use in working with applications compatible with the Digidesign product line. It lets one store samples and their effects parameters in two separate zones. This optimizes the time necessary for editing (changing the selection of samples) and saving (writing samples to disk).*

• ***Audio Interchange File Format (AIFF)*** *– a standard exchange format defined by Apple Computer and used by many music applications for different purposes. This compatibility is an advantage when one wants to put a file through multiple transformations using different programs.*

Hyperprism also reads files in **Hyperprism document format**. As Chapter 3 shows, Hyperprism files do not contain sound samples. Rather, they contain effects parameter settings and links to SD II and AIFF sound files.

Hyperprism also generates **pitch and envelope analysis files** used with the Pitch and Envelope Follower effects. These are described in Chapter 4 and Appendix B.

Hyperprism does not support the old monaural Sound Designer I format.

3

Guided tour of Hyperprism

Once you have installed and configured the program, as described in Appendix A, you are ready to trek through the world of Hyperprism. You needn't follow every step of this tour; if you like, you can skip around. Here are the basic guideposts of the tour.

- a right pointing arrow (⇒) invites you to execute a command with the mouse or keyboard.
- a down pointing arrow (↘) describes the result produced by executing a command.
- the instruction "Command" means type the Apple (⌘) key, as in "Command-Q" for Quit.
- a pointing hand (☞) indicates an important note.

☞ If you experience any problems in the course of this tour, refer to the Troubleshooting section in Appendix C.

This tour takes you through the following steps:

Starting Hyperprism

Immediate Gratification

Playback Palette

Peak Meters

Faders

Relation Between Hyperprism Documents and Sound Files

Interpreting the Document Reference Window

Processing an Example Sound File in the Blue Window

Opening a Blue Window

Adjusting Settings

Applying an Additional Effect Process

Selecting, Deleting and Saving Variations

Multipass Processing

Processing Your Own Sound Files

Compatibility Between Hyperprism and the Sound Designer II Application

Using Thru Mode

Starting Hyperprism

After having installed Hyperprism and copied the example files onto your hard disk as explained in Appendix A, start Hyperprism as follows:

Double-click on its icon.

Hyperprism appears and displays three tool palette windows (Playback, Faders, and Peak Meters).

Choose New in the File menu or type its keyboard equivalent Command-N.

A document selection window appears, showing the contents of your current disk. Locate the folder into which you copied the sound files examples from the diskettes provided in your package. To follow this tour, choose the sound file named Beethoven.

Double-click on Beethoven (or click it once and select Open).

The screen displays the document reference window for the file Beethoven.

Click on the Play button at the left of your screen.

The Beethoven sound file plays while the Peak Meters indicate the sound level. You may adjust the sound level by clicking and dragging on the Faders control.

Now you are set to work on this file. This mode of operation is called *File mode*, in contrast to *Thru mode* in which you work with sounds coming directly into and out of your Macintosh. Thru mode is covered at the end of this chapter.

Immediate Gratification

☞ If you have more than one sound card, you need to configure Hyperprism before using it (see Appendix A).

Hyperprism is easy to use. If you have just one Digidesign card and you would like to hear what it does without further ado (assuming you have already created a new Hyperprism document for the file Beethoven), follow the instructions below:

➠ Select Pitch Time Changer in the Processes menu.

↘ A Blue Window appears, corresponding to the chosen effect.

➠ Starting from the center point, hold the mouse button down and move the tracer around in the plane (play zone) of the Blue Window.

↘ As you draw with the tracer, you'll hear the result of varying the parameters of the effect in real time.

➠ When the sound stops, click on the Play button to play back the effect you created.

↘ Hyperprism remembers your movements and retraces them automatically.

If you like, try other effects immediately by selecting them from the Processes menu (except for Pitch and Envelope Followers which are special and are covered later). As you learn more about Hyperprism, you will discover the power and flexibility of its features.

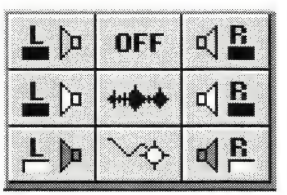
Playback Palette

After experimenting with your first effects, you should close the Blue Windows to continue with this Guided tour. Keep only the document reference window named Beethoven-FX on your screen.

The Playback palette appears in the upper left corner of the screen when you launch Hyperprism. It contains two parts: listening mode controls and tape transport controls.

Listening Mode Controls

The top part of the palette contains nine elements (three rows and three columns) which let you select the listening mode. The green lights indicate which listening mode is currently active. From top to bottom, the three rows represent *mute mode*, *original mode* and *processed mode*. You can independently select the left or right channel, or select both channels by clicking on the center column.



Listening modes in the Playback palette

Click on the center button (OFF) on the top row.

The simulated green light-emitting diodes (LEDs) in the top row light up.

Click on the center button (yellow point) in the third row.

The green LEDs in the top row go out and the LEDs in the bottom row light up.

In this mode you hear the sound *while* it is processed or *after* it has been processed by an effect.

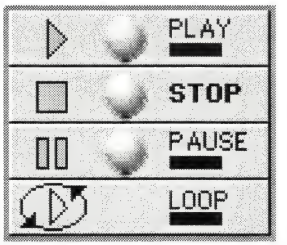
Most effects let you choose a different listening mode for each channel. Just click on one channel at a time. For example, you could be listening to the original sound in the left channel and the processed sound in the right channel. (The Vibrato, Doppler and Pitch Time Changer effects do not support either this type of left/right comparison listening or switching from *Original* to *Processed* mode.)

Click on the center button in the middle row to activate the original sound listening mode. We'll need this mode in a minute.

The green lights in the center row light up.

Tape Transport Controls

The lower part of the Playback palette, containing four items, lets you select the tape transport mode. It groups together the traditional transport controls of a tape recorder.



Tape transport controls in the Playback palette

Play
(or Space-Bar)

Now let's listen to our example file Beethoven.

Click on Play.

The green Play LED lights up. It goes out at the end of the sound. You hear the original sound in the right and left channels.

Pressing the Space-Bar is equivalent to clicking on PLAY.

Stop
(or Return key)

Click again on Play and, after a couple of seconds,

Click on Stop.

The green Play LED goes out and the playback stops.

If you click again on Play, Hyperprism starts at the beginning of the sound.

☞ Pressing the Return key is equivalent to clicking on Stop.

Pause (or Space-Bar)

➠ Click again on Play and, after one or two seconds,

➠ Click on Pause.

➘ The green Play LED remains lit while the Pause button is engaged. To continue to hear the sound,

➠ Click again on Pause or on Play.

➘ The green Pause LED goes out and the sound continues to play.

☞ As an alternative to the mouse, the Space-Bar can also be used to start and pause playback. Hit it once to start and again to pause.

Loop

The Loop function lets you listen to the sound as it plays repeatedly. It functions with the Play button, and must be activated with it. To hear our file Beethoven in a loop,

➠ Click on Loop,

➘ The green Loop LED lights up.

Click on Play.

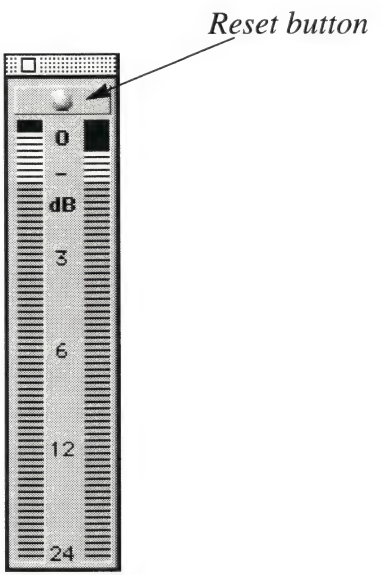
The green Play LED lights up and the sound will continue to loop until you stop it.

You can also click on Loop while a sound is already playing.

To deactivate the Loop function, just click again on Loop.

Peak Meters

The peak meters indicate the dynamic amplitude level of the sound as it plays.



Peak meters

Hyperprism's meters inform you of possible overloads in the signal level. In the upper part of the scale, orange LEDs light up to indicate that the level is approaching the saturation

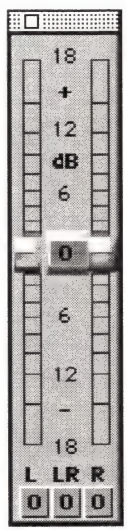
point. Red LEDs at the top indicate saturation. The red LEDs remain lit even if playback is interrupted by clicking on Pause or Stop. To turn the red LEDs off, click on the *reset button* in the top center of the peak meter window.

☞ When an effect is on, Hyperprism's meters indicate the level at the output of the effect (i.e., post-effect).

To set the output level, you can adjust the faders as described below.

Faders

The faders are graphic potentiometers that produce a gain or attenuation of the output signal. They are located to the right of the peak meters by default.



Faders

Each channel is independent. The fader palette contains:

Two indicator columns (left and right) graduated from –18 dB to + 18 dB.

Two faders, left and right. These can be independent or grouped together. The faders are aligned to 0 dB by default. To move both faders simultaneously, click the knob in the middle and drag the mouse up or down. You can also select the individual channels if you wish.

Three buttons at the bottom of the window each show 0. Click on one of them to reset the faders to 0 (left, right, or left and right simultaneously).

☞ You can change the fader levels during playback. If you want to change the gain, be prudent when increasing the level, as you might overload ears and your loudspeakers. If you saturate the digital-to-analog converter hardware, the red LEDs in the peak meters light up. They remain lit until you explicitly turn them off with the reset button at the top of the meters.

Pre-effect and Post-effect Level Adjustments

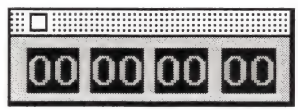
When an effect is on, Hyperprism's faders affect the level at the output of the effects process (i.e., post-effect).

You can adjust the level of the signal going into the delay and echo effects by using the *Pre Attenuation* setting in the Settings... dialog under the Processes menu.

In Thru mode, you may adjust the input level in the Digidesign card window accessible via the Setup menu or by typing Command-D.

Time Display

The Time Display displays run time as a sound is played back or while a path is recorded or played.



Time Display Palette

Relation Between Hyperprism Documents and Sound Files

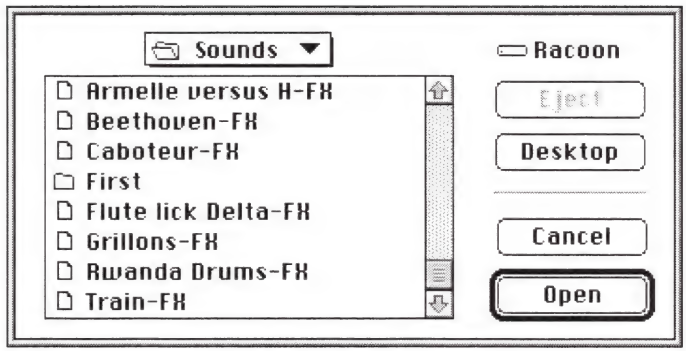
☞ *In Hyperprism, sound files are never open directly. When in file mode you always either Open or create (New) Hyperprism documents.*

Hyperprism has its own files called *documents* for purposes of efficiency. (The term "document" is a Macintosh system term referring to any file created by an application.) A Hyperprism document is NOT a sound file because it does not contain the sound samples.

Icon of a Sound Designer II File (left)
Icon of a Hyperprism Document (right)

Open

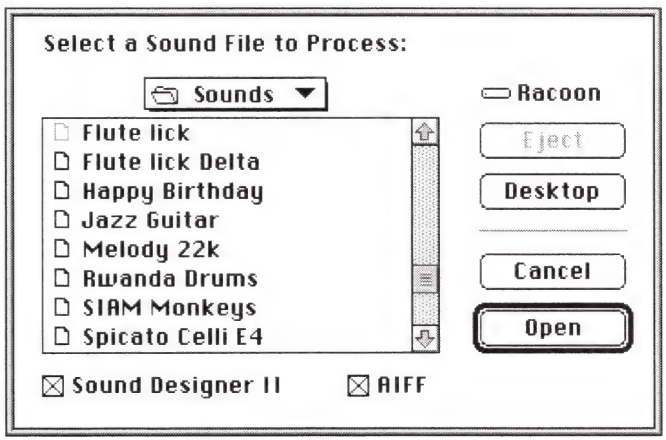
When you select Open in the file menu, notice that none of your sound files appear in the selection menu. This is because Hyperprism Opens only Hyperprism documents which are created with the New command.



The Open Dialog

New

To begin processing a sound file, select New in the File menu. This operation creates a new Hyperprism document:



The New dialog

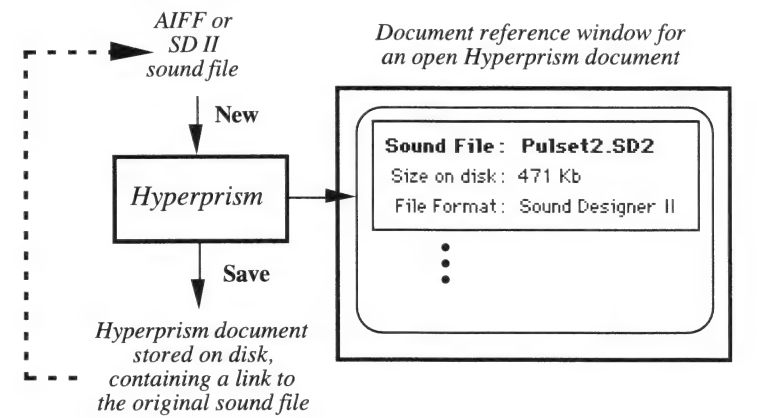
Note that you can decide which format of sound files to display by checking on the corresponding check box, either SD II and/or AIFF.

Save

When you select Save in the File menu, only the Hyperprism document is saved. It contains the parameter values you used in processing the sound input file and also contains a link to the original sound file, which is untouched.

☞ **Hyperprism is nondestructive**: *it never alters the original sound file.*

The process of sound file processing with Hyperprism can be depicted as follows:



Hyperprism documents on disk are linked to the original sound file. The dotted line represents the link between the Hyperprism document and the sound file.

The link between a Hyperprism document and the original sound file is handled by Apple's powerful Alias Manager (the same System function is used when you make an alias

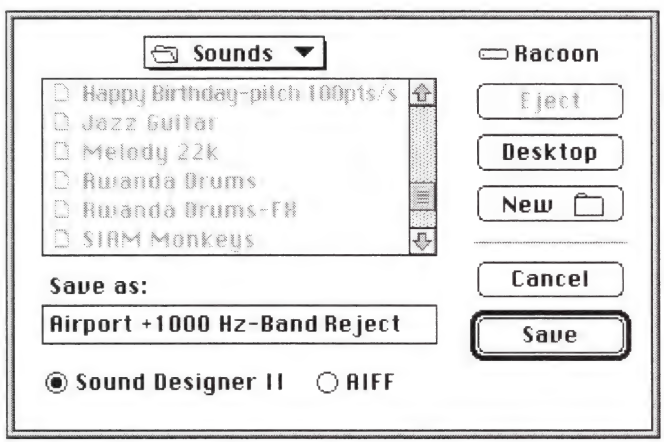
from the Finder). This means that you have a great deal of freedom to move or rename both files while maintaining the link between them. You can, however, deliberately change the link by using the Reassign command.

Reassign

You can link a Hyperprism document with its variations to a sound file other than the original one for which it was created by using the Reassign command in the File menu. To avoid confusion when Reassigning Hyperprism documents, we recommend that you rename them or create copies (using Save As or Save A Copy in the File menu) to indicate the appropriate link to the input sound file.

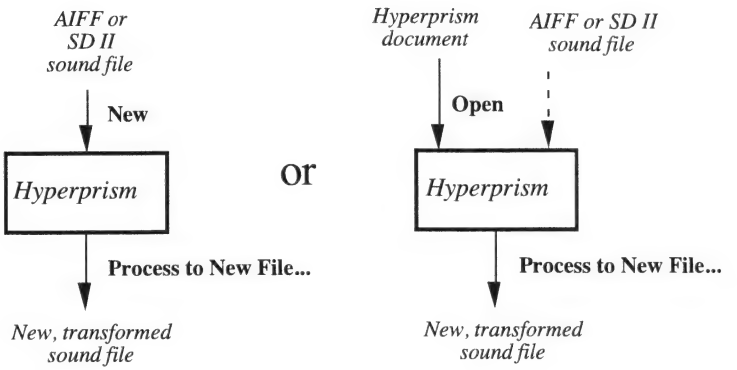
Process to New file

You can also use Hyperprism to create new sound files using the Process to New File command in the File menu.



The Process to New File dialog

The diagram below depicts the Process to New File:



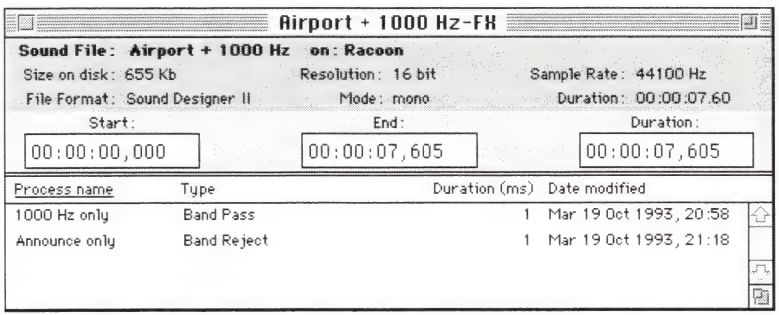
Creating a new transformed sound file, starting from an unprocessed sound file (left) or an existing Hyperprism document and the sound file linked to it (right)

As the diagram shows, you can create a new sound file in either of two ways, starting from an unprocessed sound file (invoked via New) or from a previously created Hyperprism document, which automatically reads its linked sound file.

☞ When generating a new sound file you have the option of changing its format from AIFF to SD II or vice-versa.

Interpreting the Hyperprism Document Reference Window

Whenever you process an input sound file, Hyperprism creates a document reference window. The Hyperprism document reference window serves as the information window for an open Hyperprism file.



Hyperprism document reference window

The top of the document reference window is a *title bar* which by default specifies the name of the original input sound file appended by the string "-FX." This helps you differentiate between a Hyperprism (-FX) document and the original sound file. Naturally, you can rename the document at your convenience in the Save As dialog or in the Finder. We recommend that you rename a Hyperprism document or make a copy with a new name (using Save A Copy in the File menu) when Reassigning it to a new sound file.

Below the title bar is a grey zone (regardless of whether you have a color or grayscale screen) containing the following general information about the sound file:

Sound file:
Name of the sound file linked to this Hyperprism document, followed by:

on:
Name of the disk on which the file resides

Size on disk:
in Kbytes

Resolution:
Number of bits per sample (only 16 in this version of Hyperprism)

Sample Rate:
44100 or 48000 (44.1 KHz or 48 KHz)

☞ If your sound file doesn't have one of these specific sample rates, you'll need to open your file in Sound Designer II and perform a SR Convert command via the DSP menu.

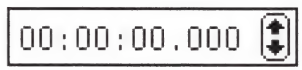
File Format:
AIFF or SD II

Mode:
Number of channels (1 or 2), i.e. Mono or Stereo

Duration:
In hours, minutes, seconds, hundredths of a second

Using the Duration Counters

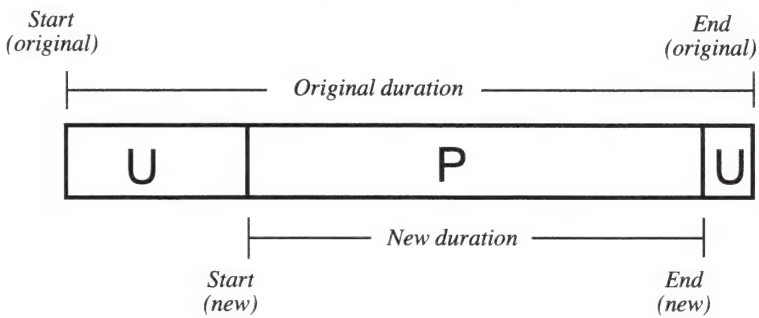
Below the file information zone in the document reference window are three duration counters named Start, Stop, and Duration. They indicate timings in hours, minutes, seconds, and milliseconds.



Duration counter

For the sound file Beethoven, the Start value is 00:00:00.000, the End value is 00:00:04.724, and the Duration is equal to the End minus the Start or 4 seconds and 724 ms.

You can modify any of these counter values in order to shift the Start time or End time, and consequently, the duration of the processed sound. This lets you isolate part of a sound file for processing, leaving the rest untouched.



Adjusting the Start and End time to process only a selected portion of a sound file. The "U" indicates the unprocessed part of the sound file. The "P" indicates the processed part.

Several methods can be used to set up the Start, Duration, and End counters.

Click in the Start field.

It becomes highlighted and two black up-down arrows appear. You can then,

Directly type a new numerical value, or

Click on one or the other arrow to change the value.

Type 1 in the seconds zone and 260 in the milliseconds zone in order to get: 00:00:01.260. Finally, following the same principle,

Change the value in the End field so it becomes 00: 00:02.140.

Notice that the duration is recalculated.
Here you have shortened the duration of the sound from both the start and the end.

Now click on Play.

You hear only the central part of the example file.

Using the up-down arrows, you can accelerate the counter by moving the mouse up (for incrementing) or down (for deincrementing) as you hold down the mouse button.

When you modify the Duration counter, this affects the End point.

☞ You can also move the cursor within the counter fields using the arrow or tab keys.

☞ It is also possible to modify the counter values in real time while hearing the sound. For best results, try this in Loop mode (with no effect selected). Once you have selected the proper portion, you can then apply an effect process to only this portion.

List of Effect Variations

Process name	Type	Duration (ms)	Date modified
Untitled BandPass	Band Pass	13717	8 Oct 1993, 11:23
Untitled Echo	Echo	5100	8 Oct 1993, 11:23
Untitled Phaser	Phaser	12266	8 Oct 1993, 11:24
Untitled Vibrato	Vibrato	9221	8 Oct 1993, 11:23

List of effect variations

When you first create a new Hyperprism document, the variations field in the document reference window is empty. Every time you select an effect, a new entry appears in this list. The list has four columns:

Process name:
listed as "Untitled" followed by an abbreviated name of the effect, for example, "Untitled PitchTimeChgr."

Process name	Type
Untitled BandPass	Band Pass
Untitled Echo	Echo
Untitled Phaser	Phaser
Untitled Vibrato	Vibrato

☞ You can customize this name by selecting Get Info… in the File menu.

Type:
Type of effect. Even if you change the Process name, the Type remains the same.

Duration(ms):
Duration of the variation in milliseconds. That is, the duration during which you have recorded points in the Blue Window. This value may not be updated until the window is redrawn (for example, after you select another window and then return to the document reference window).

Process name	Type
Untitled BandPass	Band Pass
Untitled Echo	Echo
Untitled Phaser	Phaser
Untitled Vibrato	Vibrato

Date modified:
Date and time that you last applied this effect.

By default, the entries are listed in alphabetical order by effect name. Notice in the figure above how "Process name" is underlined. As in the Macintosh Finder, you can reorder the list according to Type, Duration, or Date by clicking on that column name. This causes the selected column name to be underlined and the list to be sorted according to the values in that column.

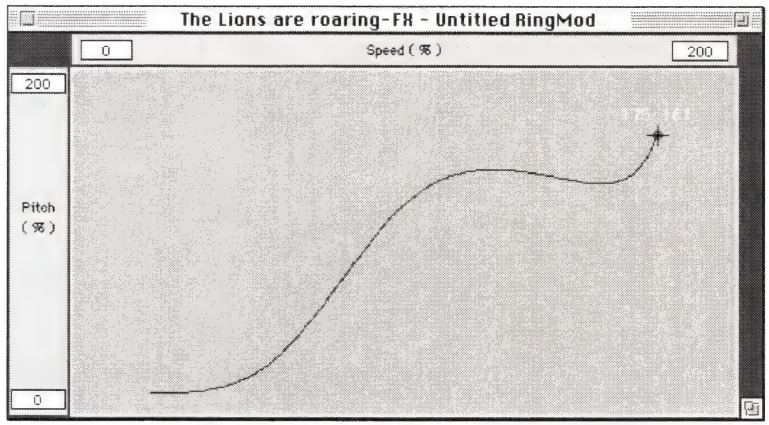
Processing an Example Sound File in the Blue Window

Let's begin to explore the world of sound transformation with Hyperprism. If you have followed the instructions so far, the window of the example sound Beethoven is open. Now it's time to choose an effect from the Processes menu. We suggest that you choose Pitch Time Changer. This powerful effect lets you independently modify the pitch and the playback speed of the sound file.

Opening a Blue Window

Click in the Processes menu and choose Pitch Time Changer.

A new, Blue Window appears and, in the listening mode section of the Playback palette, the green LED indicators automatically switch to the bottom row to indicate processed listening mode.



A Blue Window™

A Blue Window has four parts:

Title bar:
Contains the name of the input file followed by the name of the chosen effect, i.e., "Beethoven-FX-Untitled PitchTimeChgr."

Horizontal ruler:
Situated under the title bar, this scale represents one effect parameter. For the Pitch Time Changer, this scale represents the degree of speed variation.

The default speed range is from 0 to 200%, where 100% means the original speed. The default parameter values can easily be changed (see “Rescaling the Play Zone” later in this chapter).

Vertical ruler:
Situated at the left of the Blue Window, this represents a second effect parameter. For the Pitch Time Changer, this represents the degree of pitch change.

The default pitch change range is from 0 to 200%, where 100% means original pitch. The default parameters values can easily be changed (see "Rescaling the Play Zone" later in this chapter).

Play zone:
It is precisely the blue part of the window. It contains a yellow tracer icon, which is initially stationary. The sound begins to play when you press the mouse button. As you press the mouse button and drag the tracer across the surface, you change the parameters of an effect algorithm.

You can also click the mouse at different points in the play zone for a stepped or abrupt change in parameters, in which case Hyperprism connects the click points with straight lines. You may click at different points in real time or you may pause (by using the Space-Bar or Pause button), click at a different point, and then exit pause to continue processing.

All your movements in the play zone are recorded by Hyperprism. In a moment we will explain how to play back a path drawn in the window.

The Blue Window is resizable via the grow box at the bottom right of the window. This feature is useful when you want more precision in manipulating the parameters. With a large screen and a large Blue Window you can realize precise control over effect parameters. Small windows can be useful for changing values over a wide range with only a small gesture. The pixel-to-parameter relation can be fine-tuned to your needs using the grow box and/or the rescaling possibilities (see "Rescaling the Play Zone" later in this chapter).

The Tracer

When the mouse cursor appears on the play zone, it changes into a *tracer* icon. As you press the mouse button and move, it traces a *path* and displays the X- and Y-axis values associated with the current point. These values are always displayed as integers in the format:

[horizontal] , [vertical]

While only integers are displayed, internally the program takes decimal values into account. If, for example, you rescale the horizontal scale to between 0 and 1, horizontal mouse movements generate internal values that are precise to several decimal places.

Tracer at horizontal 1 and vertical 9

The tracer icon takes three forms:

Red: (or black if you have a monochrome monitor)
When the effect is started by clicking in the play zone and at least one point is recorded.

Transparent:
When the mouse button is not pressed. You can move around the play zone without affecting the sound, positioning the mouse at a precise point.

Yellow: (or white if you have a monochrome monitor)
Before or after you have applied an effect and when you click on the Play button to hear it retrace the path you previously drew.

Playing Back a Path

Since all your movements are recorded, Hyperprism can play back what you've done: just click on Play or press the Space-Bar to play back the path. If you click and drag twice in a row, processing a sound file twice with different parameter settings, you'll see that Hyperprism draws a straight line between the two performances and plays them consecutively when you click on the Play button.

Deleting a Path

To delete a path drawn in the Blue Window, select Clear in the Edit menu or press Backspace or Delete on the keyboard.

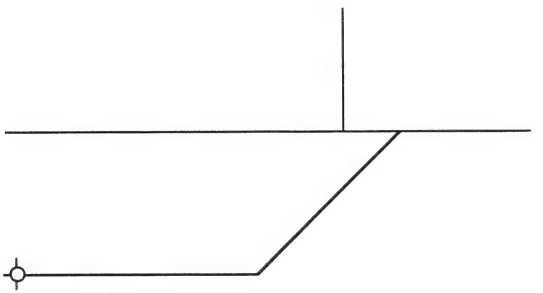
☞ You can delete a path while recording or playing it back. This will stop the sound and clear the Blue Window. You can also delete a path directly from the reference window, by selecting its corresponding line and pressing Backspace. If its Blue Window is open, it will disappear.

Looping a Path

The Loop listening mode functions in a special way. Say you process an effect in Loop mode, drawing a path over four iterations of a sound file. When you play back this path, also in Loop mode, Hyperprism will play the path over the four iterations repeatedly, looping the original loop.

Tracing Straight Lines in the Play Zone

Sometimes you may want to vary just one of the parameters, while leaving the other fixed. Press and hold down the Shift key. Depending on the direction you move the mouse, the tracer follows a straight horizontal, vertical, or diagonal (45 degree) line.



Tracing straight horizontal, vertical and diagonal lines in the play zone

Another method to create a path with one fixed parameter is to rescale the play zone.

Rescaling the Play Zone

The default parameter limits that appear in the corners of the Blue Window are just a starting point for sound transformation. Only occasionally will they be the ideal settings for a given input sound. Hence, the ability to rescale the parameter limits is one of the most important features of Hyperprism. This is where you tune an algorithm to the qualities of a particular sound or to the shape of a particular gesture made with the mouse.

You have several ways to rescale the parameters of the Blue Window. The first is to simply click on one of the limits and type in a new value. Another way to select parameter limits is by hitting the tab key. As you hit tab several times, watch as it moves around the parts of the window, highlighting each in turn. If you modify the limits of a parameter and then move the tracer in the play zone without pressing down the mouse button, the tracer does not reflect the new parameter limits. In order to "update" the play zone and the tracer you must either:

- Hit the tab key until the play zone is selected, or
- Click in the play zone to activate it, which will also start the effect.

Now let's give some examples of rescaling. Let's suppose you want to vary only speed with the Pitch Time Changer while keeping the pitch constant. Here you could set both the upper and lower pitch limits to 100%. Then no matter what path you would draw in the play zone, the pitch would remain constant. Or open up the pitch change to a semitone range (about 6% variation) by setting the upper limit to 103 and the lower limit to 97%.

Rescaling the parameter limits means that you can hone in on a particularly effective region for a given input sound. Reducing the range of the frequency cutoff in a filter, for example, often leads to more controllable and interesting effects. Try a Band Pass Filter with a range between 20 and 1000 Hz in the frequency scale (horizontal) and 0 to 15 in the *Q* scale (vertical). (See also the Filter examples in chapter 4).

☞ To revert to default parameters, open a new Blue Window with the same Process name.

Parameter Resolution Tradeoffs

The screen of your Macintosh has fundamental limits in the number of pixels it displays, which limits the resolution of the values you can specify using a mouse. If, for example, your play zone is 1000 pixels wide and the chosen effect is a filter with a frequency range from 20 to 20,000 Hz, notice that you have only 1000 pixels of addressing into a space that is 19980 values wide. This means that if you nudge the tracer by one pixel, the value jumps by 19 Hz. To increase the resolution of the tracer you can either increase the size of the play zone (large screens help here) or reduce the range of the parameter limits. If you reduce the frequency bandwidth of the above window to 1000 Hz, for example, you attain 1 Hz resolution on the screen.

Effects Limitations

Each effect in Hyperprism has certain limits determined by the speed of the underlying hardware. For example, in the Pitch Time Changer, it is not possible to increase the playback speed beyond 200%. Also, no matter what effect is chosen, the four parameters of the Blue Window are limited to take values only between -32768 and +32767. Not all values in this range are meaningful for all effects. In some cases, like the Pitch Time Changer, increasing the value beyond 200% has no audible effect, nor does a filter cutoff greater than 20 KHz.

Adjusting Settings

Besides the parameters shown in the Blue Window, you can also adjust one or more parameters in the Settings... window associated with each effect. To see the Settings window for a particular effect, select an effect first and then select Settings... in the Processes menu. Settings are discussed in the second part of Chapter 4.

Applying an Additional Effect Process

Hyperprism lets you apply up to two effects simultaneously. You can apply Single Delay, Multi Delay, or Echo to any open effect in File mode. (However, you cannot apply an additional effect to the Single Delay, Multi Delay, or Echo themselves.) The reason for this is that the delays are created by the Macintosh CPU while the other effects are created by the Digidesign DSP.

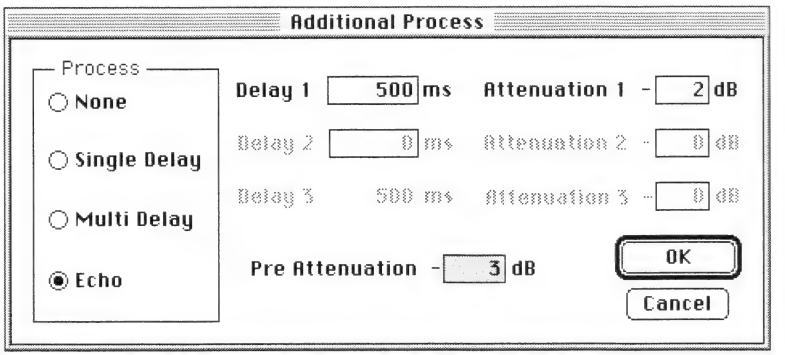
To do this, assuming your Hyperprism document Beethoven-FX is open,

➠ Select a new Pitch Time Changer in the Processes menu or delete the path of one that is already open. Then,

➠ Click in the Processes menu and choose Additional Process....

➘ A new window appears in which you can choose the additional delay process and specify its parameters.

➠ Click on the Echo button. Then, in the Delay 1 field at the top of the window,



Additional Processes... window

- Type the value of the delay parameter that you would like, for example, 500 (milliseconds).
- Then type the Attenuation 1 value, for example, 2 (the amount by which the first echo is attenuated in dB). You can leave the value set at the default of 0 dB if you wish, but realize that no attenuation means that the echo will repeat indefinitely.
- At the bottom of the window is a field for Pre Attenuation, which is the amount that the input signal is attenuated before being processed. A -3 to -9 dB Pre attenuation factor is recommended for loud sounds fed into delay and echo effects in order to avoid clipping. In this case, since the sound file has a strong dynamic, it is wise to set the pre attenuation factor to 2 or 3 (meaning -2 or -3 dB).

Click on the OK button.

The Additional Process... window disappears, while the Blue Window (Beethoven-FX-Untitled PitchTimeChgr) becomes active. You can hear the effect with the Echo by clicking in the play zone and moving the mouse around.

Notice the effect of the echo itself while speed and pitch vary.

It is also possible to open an additional process when a path has already been created.

The Additional Process works in two modes, *pre-path* and *post-path:*

Pre-path

If you select the Additional Process... before you draw a path, then the total sound duration will continue until all delays and echoes die out.

Post-path

If you draw a path and then select an Additional Process..., this guarantees that the total sound will end at the end of this path.

In this case, the truncation of the delay effect selected in the Additional Process... window is normal.

Cancelling an Additional Effect Process

To cancel the Additional Process:

➠ Click on the Processes menu and choose Additional Process....

↘ Its window appears. Then,

➠ Click on the button "None" in the Process frame.

↘ All the values are grayed out, and thus become invalid.

➠ Click OK.

Selecting, Deleting, Undoing, and Saving Variations

As mentioned earlier, a *variation* is an effect applied to an input sound file. Each variation is listed in the document reference window. Closing a Blue Window does not delete its entry from the variations list. All you have to do to bring it into view again is

➠ Double-click on its entry in the variations list or

➠ Click on its entry in the variations list and click Display Process Window in the Processes menu.

➘ The Blue Window containing the variation reappears.

If you are not pleased with the variation, you don't have to keep it. To delete a variation:

➠ Click on the corresponding line in the variations list and then select Clear from the Edit menu (or press Delete or Backspace).

➘ The entry in the document reference window disappears, as does the Blue Window associated with that variation.

To undo a deletion, go to the Edit menu and select Undo or type Command-Z.

If you like the variation you've created, you can save it along with other variations in the Hyperprism document. All variations are stored together. To save your variations in a Hyperprism document:

➠ Click on the Close button on the window, the system will ask you whether you want to save the document or not, or,

➠ Select Save... in the File menu, or,

➠ Select Save As... in the File menu to rename the file, or

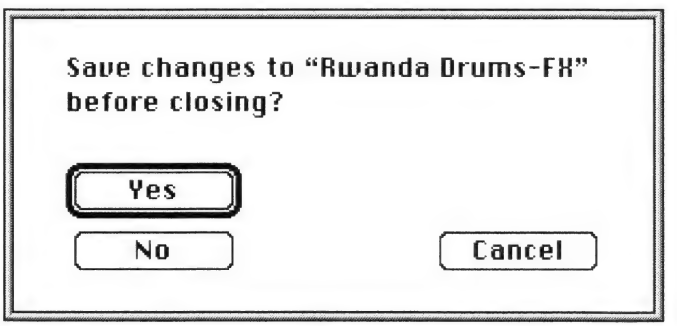
➠ Select Save a Copy... in the File menu to save a copy that is automatically renamed.

☞ Saving a Hyperprism file saves only the effect parameters. This is generally a very small disk file, and the original sound file is left intact.

Hyperprism document icon

If you quit the application without having saved the Hyperprism document containing variations, a save option will be automatically proposed to you.

You can click in the button of your choice or use the keyboard equivalents Y, N or Esc.



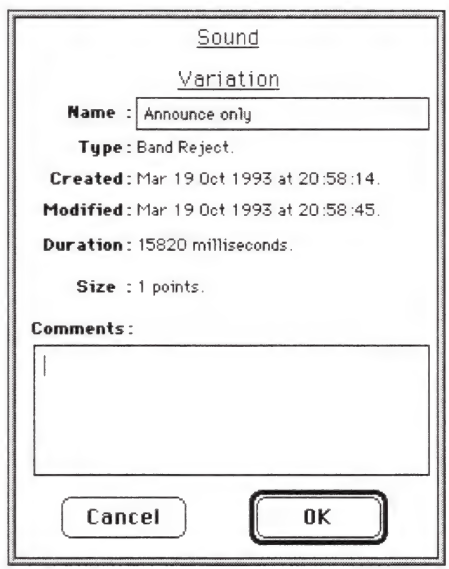
Automatic Save dialog for Hyperprism document

Customizing the Process Name and Supplying Comments

Customizing the process name is a good way of keeping track of the nature of the variation. To change the process name, you must have the corresponding Blue Window open in front or its line in the document reference window selected.

Select Get Info... from the file Menu or type Command-I.

A new window appears, describing the variation. Type the new name.



Info box for an effect variation

You can also write comments. These comments pertain to a specific variation, not to the entire file, so when you look in the Info box for the entire file, you won't see these comments. They appear only in the Info box of a particular variation.

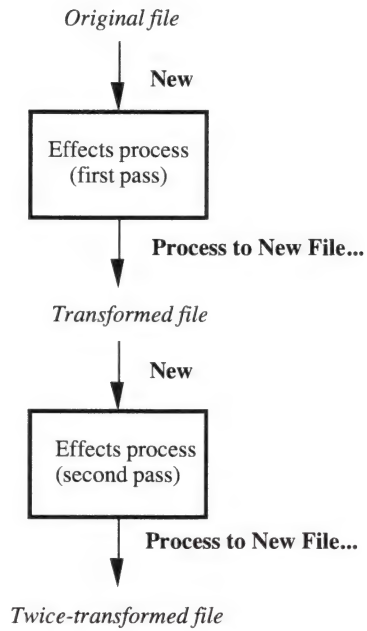
When you are done typing comments, click OK.

The Get Info... window disappears and the document reference window reappears with the new variation name.

Once you've made one variation, more possibilities open up. For example, you may want to apply another effect on top of a previously created one. This procedure is described in the following section.

Multipass Processing

Hyperprism is a multipass effects processor. To apply a series of different effects, you generate a new sound file after each pass through a single effect. In this way you can apply dozens of different, simultaneously sounding effects to the same file. After each pass through the effect, select Process to New File... from the File menu. This creates a new sound file. Then you create a new Hyperprism document for the newly-generated sound file and run it through another effect, and so on.



Multipass effects processing

Multipass processing also lets you exceed the limitations of an individual Hyperprism effect. Each effect in Hyperprism is limited by the speed of the underlying hardware. For example, in the Pitch Time Changer, it is not possible to increase the playback speed beyond 200%. The workaround to these kinds of limitations is easy; simply carry out the operation in more than one pass. Treat the file to the maximum allowed in one pass and save the result with Process to New File..., then create a new document containing the just-processed file and apply the same effect.

☞ The length of the new file is related to the duration of the Process.

Processing to New File

Say that you want to apply a Tremolo on top of the Pitch Time Changer + Echo you already created.

Bring the corresponding Blue Window to the front.

Choose Process to New File... in the File menu.

A save window appears. You are about to generate a new sound file. The default name is taken from the document reference window, i.e., the name of the original file appended with the name of the effect. To change this name,

Type the name of your choice or for the sake of this example, just delete the words "FX-Untitled" and click on Save.

A thermometer appears. It lets you estimate the time that the operation will require. You can interrupt the operation by clicking on Stop.

The duration of the generation operation has been optimized to last slightly over twice the duration of the sound itself.

When the operation is finished:

Select New and create a new Hyperprism document for the newly generated file "Beethoven-PitchTimeChgr."

A new document reference window appears. Note that this window has no effects variation line, since it is new.

Click in the Processes menu and choose Tremolo.

A Blue Window appears, called "Beethoven-PitchTimeChgr-Untitled Tremolo."

The file is ready to be processed again. To do this,

Click and move the mouse within the play zone.

You hear the original sound Beethoven affected simultaneously by a Pitch Time Changer, an Echo and a Tremolo.

This kind of multipass processing can be continued indefinitely, with any combination of effects, as long you have sufficient disk space to hold all the resulting sound files.

The new file is usually about the same size as the original, except in the case of the Pitch Time Changer, which can stretch or shrink the duration. This applies also to delay effects, whether they are used as varying processes or as additional, fixed processes.

Thus, Hyperprism offers two alternative strategies for sound processing:

❶ Creating many variations of the same input signal, applying the same or different effects. This strategy uses little additional disk space, since only the effect parameter values are saved.

❷ Creating several processed sound files, starting from the original file. This takes up more disk space, but it also offers the option of multipass processing (applying several effects to the same input sound). To save space, you can always delete previous results as you go. Note that as long as you keep the related Hyperprism documents, you can always regenerate any of the intermediate sound files by selecting the proper variation and doing the Process to New File... again.

Processing Your Own Sound Files

Hyperprism lets you process sound files in either the AIFF or SD II formats. Sound Designer II records files in SD II format. To see how to record new sounds, read the documentation supplied by Digidesign. Once you've recorded your sound with Sound Designer II, all you need to do is:

➠ Select New from the File menu and then select your sound file in the dialog box that appears.

➘ Choose an effect from the Processes menu.

Other sources for sound files include programs such as Digidesign's TurboSynth, a software synthesis program.

Compatibility Between Hyperprism and Sound Designer II

The compatibility of Hyperprism with Sound Designer II is subject to certain limitations:

- Old Sound Designer I (monaural) format is not supported.
- Only 16 bit recordings are supported.
- The sampling frequencies supported by Hyperprism are 44.1 and 48 KHz.

☞ Hyperprism and Sound Designer II can both be open at the same time, but the two applications cannot simultaneously open the same file. You must thus take this into account in working with both programs at the same time, perhaps saving and closing from Sound Designer II, then reading into Hyperprism, or vice versa.

☞ We have found that the Alchemy sound editor by Passport Designs can open the same file simultaneously, but that the two versions are not synchronized. Thus if you modify the file with Alchemy and save it, Hyperprism will not recognize that version until it closes and reopens the document linked to that file.

Using Thru Mode

Besides processing files, Hyperprism can process sounds coming in directly through the digital or analog audio input jacks of the Digidesign card. In this "live" Thru mode, your Macintosh behaves like any standard effects processor. You hook it up to a sound source and can process any sound, no matter how long, in real time.

Effects Available in Thru Mode

Processes menu in Thru mode

In Thru mode, certain items are grayed-out in the Processes menu, indicating that they are not available. These effects include: Pitch Follower, Envelope Follower, Single Delay, Multi Delay, and Echo.

Starting Thru Mode

With no document reference window open, Hyperprism is already in Thru mode. Only when you open a document reference window does Hyperprism switch to File mode. However, if you then open the Setup Device... window (under the Views menu), Hyperprism reverts to Thru mode. As long as the Setup Device... window is open and frontmost, Hyperprism remains in Thru mode, even if you also have document reference windows open.

Configuring for Thru Mode

To use Thru mode, you may have to configure the Digidesign hardware you intend to use for input and output. If you have several Digidesign cards, you must specify which one you intend to use and its parameters (analog or digital input, sampling frequency, etc.). The bulk of this procedure is equivalent to the Hardware Setup procedure in Sound Designer II. If necessary, see the Digidesign manual.

To select the type of hardware:

Choose Select Device... from the Setup menu.

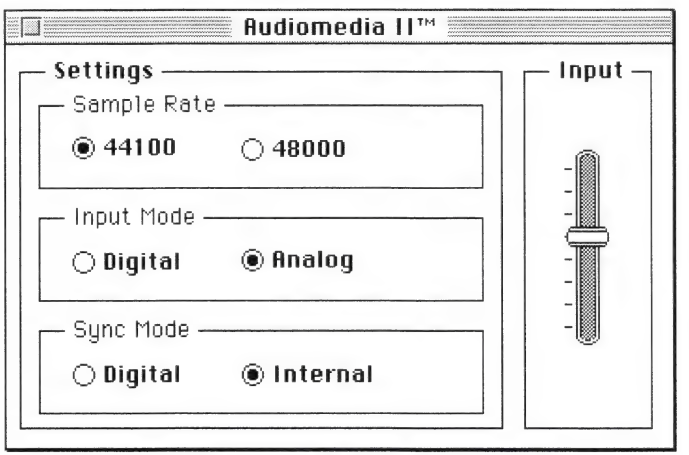
A dialog proposes a list including one or more cards installed in your Macintosh.

Select a card, click on OK.

To configure the selected device and start Thru mode:

Choose Setup Device... from the Setup menu or type Command-D.

A window specific to the selected device appears (this window opens automatically if you select a different card with Select Device...). You can adjust the available parameters.



Setup Device... window for the AudioMedia II card

As long as this window is frontmost, you are operating in Thru mode. Notice that the Setup Device... window shows up in the Views menu. You should be able to hear the unprocessed input sound source (the sound going through Hyperprism as is) by clicking the Play button in the Playback palette. If you do not hear your sound or see LEDs light up in the Peak Meters palette, refer to Appendix C.

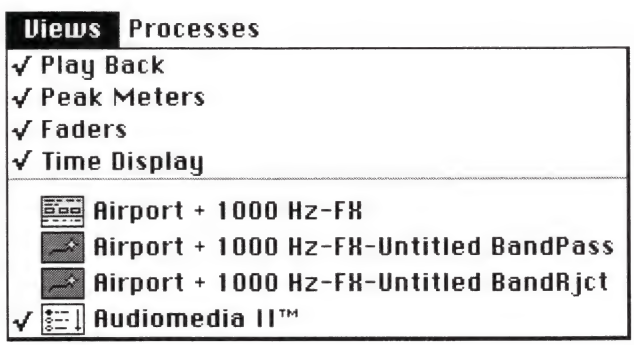
Processing a Sound in Thru Mode

To process a sound in Thru mode,

- Return to the Setup Device... window if it is not already at top level.
- Select the desired effect in the Processes menu.
- Start the processing of the live input source by clicking in the Blue Window.

Switching Between Thru Mode and File Mode

You can simultaneously open a Setup Device... window and several file windows. You can switch between them at will, alternating between File mode and Thru mode as you do.



Views menu showing the presence of a Select Device... window (for the Audiomedia II)

4

Description of Hyperprism Processes

This chapter contains two sections. The first and longest is a detailed description of each of the effects realized by Hyperprism.

The last section describes *settings* that you can apply to the effects. Settings are parametric controls beyond those associated with the horizontal and vertical dimensions of the Blue Window.

Low Pass

A low pass filter passes only low frequencies. An important property of a low pass filter is its *cutoff frequency*. By convention, this is the point in the frequency range at which the filter reduces the signal to -3 dB of its original power. Spectral components that are attenuated below the cutoff of a filter are said to be in the *stopband* of a filter. Those above the half-power point are said to be in the *passband* of the filter.

In an ideally sharp filter, the cutoff frequency is a kind of brick wall: anything outside it is maximally attenuated, dividing the frequency response neatly into a passband and a stopband. In actual filters, the slope of the filter leading up to the cutoff frequency is not linear (there is a slight *ripple* in the frequency response), and the area between the passband and the stopband is called the *transition band*.

The steepness of a filter's slope is usually specified in terms of decibels of attenuation or boost per octave, abbreviated "dB/octave." For example, a 6 dB/octave slope on a low pass filter makes a smooth attenuation (or *rolloff),* while a 90 dB/octave slope makes a sharp cutoff. The use of a smooth or sharp slope depends on the musical situation. For example, a sharp band reject filter might be needed to completely eliminate a tone centered at a particular frequency, while a gentle low pass filter could be the most unobtrusive way of attenuating background noise in the high frequency range.

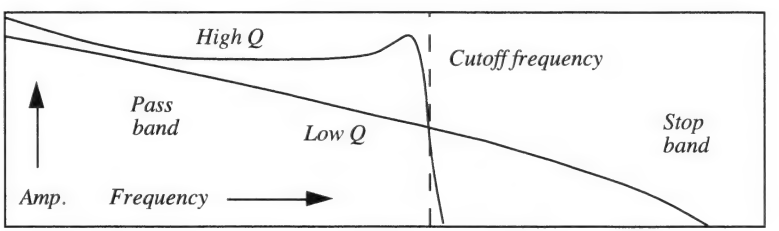
Parameters

↔ Horizontal position:

Frequency scale, by default 20 Hz to 20 KHz. Select the cutoff frequency within this band.

↕ Vertical position:

Q factor, by default 0 to 10. The *Q* determines the sharpness of the response of the filter, corresponding to its degree of *resonance*. The greater the *Q*, the sharper the filter. When the *Q* is high, the frequency response is sharply focused around a peak (resonant) frequency. A high *Q* setting will tend to reduce the total signal level coming out of the filter.



Low pass filter response curves for two different values of Q. A low Q filter has a smooth roll off. The high Q filter has a resonance peak before the cutoff frequency.

Example

Create a new Hyperprism document for the sound file Provence Ambience and play it. You should hear crickets while a propeller plane passes in the sky.

Click on Loop.

Select the Low Pass filter from the Processes menu.

A Blue Window appears.

Change the 20,000 value in the horizontal ruler of the Blue Window to 2000 in order to get a more accurate definition on the horizontal axis.

Tab until you highlight the play zone itself.

Click at the location 450,5. This corresponds to a cutoff frequency of 450 Hz with *Q* factor of 5.

You should be hearing the plane only. Note that if you drag the mouse away from this point, you'll hear the crickets coming in.

Note that the overall level has gone down. You can use the Faders to amplify the signal.

IMPORTANT

To avoid sending too loud a signal to your amplifier, be very careful to remember to put the Faders back to the middle neutral position before playing another sound or switching the listening mode to the original sound.

High Pass

A high pass filter only passes high frequencies. That is, it attenuates frequencies below the specified cutoff frequency. (For more on general properties of filters, see the description of the Low Pass filter.)

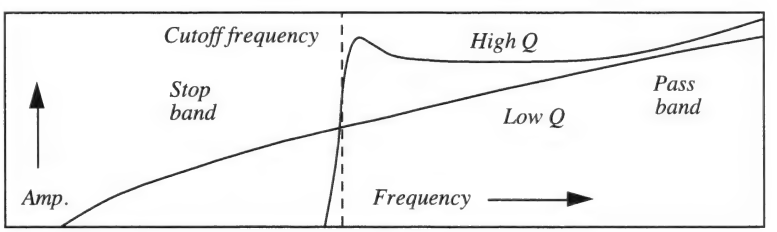
Parameters

↔ Horizontal position:

Frequency scale, by default 20 to 20 KHz.

↕ Vertical position:

Q factor, by default 0 to 10. (See the explanation of *Q* for the low pass filter.)



High pass filter response curves for two different values of Q. A low Q filter has a smooth roll off. The high Q filter has a resonance peak before the cutoff frequency.

Example

Create a new Hyperprism document for the sound file Provence Ambience and play it. You should hear crickets while a propeller plane passes in the sky.

Click on Loop.

Select the High Pass filter from the Processes menu.

A Blue Window appears.

Change the 20,000 value in the horizontal ruler of the Blue Window to 5000 in order to get a more accurate definition on the horizontal axis.

Tab until you highlight the play zone itself.

Click and drag the mouse in the play zone.

As you move the cutoff frequency from low to high, the low frequencies are attenuated more and more until you stop hearing the sound of the plane (mouse position approximately >2000,<5).

Note that the overall level has gone down. You can use the Faders to amplify the signal.

IMPORTANT

To avoid sending too loud a signal to your amplifier, be very careful to remember to put the Faders back to the middle neutral position before playing another sound or switching the listening mode to the original sound.

Band Pass

A band pass filter passes a certain band of frequencies within an upper bound and a lower bound. Frequencies above and below this band are attenuated. The difference between the higher and lower cutoff frequencies in a band pass filter is the bandwidth of the filter. The center frequency of a band pass filter is the maximum point of amplitude.

Parameters

Horizontal position:

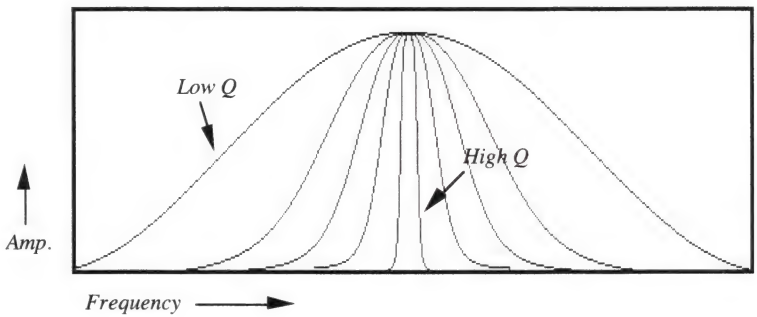
Frequency range, by default from 20 Hz to 20 KHz.

Vertical position:

Q factor, by default from 0 to 10. It determines the sharpness of the filter, and is akin to the resonance factor. *Q* can be defined precisely for a band pass filter as the ratio of the center frequency to the spread of its -3 dB point (cutoff point) bandwidth.

$$Q = \frac{f_{\text{center}}}{f_{\text{highcutoff}} - f_{\text{lowcutoff}}}$$

where f_{center} is the filter's center frequency, $f_{\text{highcutoff}}$ is the upper -3 dB point, and $f_{\text{lowcutoff}}$ is the lower -3 dB point. When the center frequency is constant, adjusting the Q is the same as adjusting the bandwidth.



Band pass filter response curve for different Q values

When the *Q* is high, the filter will begin to resonate or ring at the center frequency. However, a high *Q* setting will also tend to reduce the overall signal level coming out of the filter. The *Q* factor can be increased to a very high value, in which case it generates a sinusoidal tone amplitude of which varies according to the input signal, but at a low level in any case.

Example

Create a new Hyperprism document for the sound file Airport+1000 Hz and play it. You should hear an announcement recorded in the Paris Orly airport that contains an additional signal at 1000 Hz. The Band Pass filter lets one isolate this frequency and eliminate everything else. In order to do this:

Click on Loop.

Select the Band Pass filter from the Processes menu.

A Blue Window appears.

➠ Modify the frequency scale to a bandwidth between 500 and 2000 Hz, and the *Q* factor from 10 to 200.

➠ Click in the middle of the window toward the top. Drag the mouse in this area until only the 1000Hz tone is heard. Your cursor should indicate a value close to 1050,200.

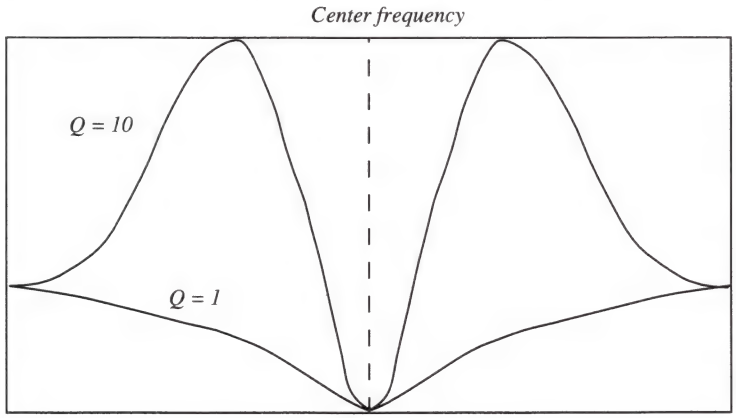
☞ Note that the overall level has gone down. You can use the Faders to amplify the signal.

☞ IMPORTANT

To avoid sending too loud a signal to your amplifier, be very careful to remember to put the Faders back to the middle neutral position before playing another sound or switching the listening mode to the original sound.

Band Reject

A band reject filter strongly attenuates frequencies within a bandwidth centered around a selected frequency. It passes frequencies above and below this bandwidth. In this sense, it is the inverse of a band pass filter, since the center frequency of a band reject filter is the minimum point of amplitude.



Approximate response curves for the Band Reject filter at a fixed center frequency, but with two different Q values. At a Q of 1, the dip in the center is smooth. At a Q of 10, two resonance peaks build up around the rejected band.

Parameters

↔ Horizontal position:

Frequency scale selects the center frequency of the rejected band. The default scale is 20 Hz to 20 KHz.

↕ Vertical position:

Q factor, by default from 0 to 10. Determines the sharpness of the cutoff of the filter. The higher the *Q*, the sharper the cutoff. See the description of *Q* for the Band Pass filter.

Example

Create a new Hyperprism document for the sound file Airport+1000 Hz and play it. You should hear an announcement recorded in the Paris Orly airport that contains an additional signal at 1000 Hz. The Band Reject filter lets one eliminate this frequency. In order to do this:

Click on Loop.

Select the Band Reject filter from the Processes menu.

A Blue Window appears.

Modify the frequency scale to a bandwidth between 900 and 1100 Hz in order to get a more accurate definition on the horizontal axis.

Click in the middle of the play zone. Drag the mouse toward the bottom until the 1000 Hz tone is not heard anymore. Your cursor should indicate a value close to 1025,0.

Note that the overall level has gone down. You can use the Faders to amplify the signal.

IMPORTANT

To avoid sending too loud a signal to your amplifier, be very careful to remember to put the Faders back to the middle neutral position before playing another sound or switching the listening mode to the original sound.

Phaser

The input signal is combined with itself after a slight, time-varying delay. This causes phase cancellation between the two signals, resulting in regular notches and peaks in the output spectrum. The shape of these peaks and notches resembles the teeth of a comb, so this type of effect is also called a *comb filter*.

Parameters

Horizontal position:

Time delay in milliseconds, by default from -5 to +5. A negative delay means that the delayed signal is subtracted from, rather than added to, the original signal, creating another type of comb filter effect which may include a sense of pitch shift. If you adjust this scale beyond 15 ms, the effect turns into a delay, in which case negative and positive delay values will make little difference in the timbre.

Vertical position:

Amplitude of the delayed signal as a percentage of the total signal, from 0 to 100%.

Example

Create a new Hyperprism document for the sound file Train. Play it in original mode and then select the Phaser in the Processes menu.

Click on Loop.

Hear the changes in the sound as you move the mouse around in the Blue Window.

Flanger

Flanger applies a time-varying delay to the input signal and then mixes the original signal with the delayed signal, creating a time-varying comb filter effect. The time-variance of the delay changes according to a low-frequency sine wave generated by Hyperprism.

Parameters

↔ Horizontal position:
Speed of the Flanger time-delay modulation.

↕ Vertical position:
Time delay in milliseconds, by default from 0 to +3. The Flanger delay oscillates (with a sine wave shape) between 0 and this value.

☞ You'll notice here that the values displayed when you move the cursor are constant, whereas in fact they can constantly change and that, between the integer values displayed, you get intermediate values which are not displayed on the screen.

Example

Create a new Hyperprism document for the sound file Train. Play it and then select a Flanger in the Processes menu.

Click on Loop.

Hear the changes in the sound as you move the mouse around in the Blue Window. If you keep the horizontal position constant you will hear the change vary sinusoidally at the speed given by the horizontal value, when for example the cursor indicates 1,3.

Chorus

A chorus effect can be heard whenever several voices sing the same note in unison. Because of slight differences in the timing, pitch, vibrato and spectrum of the different voices, we hear an ensemble effect. Hyperprism can apply a chorus effect to any input sound.

Parameters

Horizontal position:

Vibrations per second, by default, from 0 to 10, which cause an undulating pitch variation in the delayed signal.

Vertical position:

Delay in milliseconds, from 0 to 50.

This effect works well in Thru mode with a voice or an acoustic instrument. You'll find the best locations for true chorus effects when you position the mouse toward the left edge of the window and move the cursor up and down.

Note that if you move toward the right you get more and more of a vibrato effect (see Vibrato in this chapter).

Single Delay

Single Delay lets one vary the length of time (*delay time*) between the original sound and the delayed sound. You can change the delay time by moving the mouse horizontally in the Blue Window. Vertical movements change the amplitude of the delayed sound.

Parameters

Horizontal position:

Delay in milliseconds. By default, from 0 to 1000 ms. This value can be changed freely depending on the amount of memory available in your Macintosh (taking into account other open applications). In the case of insufficient memory, an alert panel may appear. You should click on "OK" and either change the delay to a lower value or allocate more memory to Hyperprism (Quit Hyperprism, do a Get Info... and increase its memory allocation size).

Vertical position:

Attenuation of the repeated signal as a percentage of the total, by default from 0 to 100%. The position at the top of the Blue Window corresponds to an attenuation of 0%, i.e., no attenuation. You may also specify a negative attenuation value which would cause the delay to be louder than the original signal.

Limitations

Delay and echo effects are sensitive to their input signals. If your input sound is full scale amplitude (i.e., normalized), choose Settings... in the Processes menu and adjust the Pre Attenuation to -3 dB or more before invoking a delay effect. Also, depending on the input signal, you may hear a clicking sound as you move the mouse in the Blue Window. To eliminate this artifact of rapidly-changing delay time, set the Parameter Change Speed (also in Settings) to 3 or higher.

Clicking sounds can also be heard if:

1. the disk is too slow,

2. the sound file is too fragmented or

3. the speed of the computer is not fast enough to handle the processing algorithms. The three delay algorithms are created by the processor of the Macintosh. It is thus essential to have a fast Macintosh for best results.

See other possible remedies in Appendix C.

Single delay is not available in Thru mode.

Multi Delay

Multi Delay delays the input signal multiple times according to the position of the mouse. The sound is repeated three times: the first time corresponding to the position of the mouse on the horizontal axis (delay1), the second time corresponding to the position of the mouse on the vertical axis (delay2) and the third time corresponding to the sum of delays 1 and 2. The attenuation values are fixed for all three delays. You can change them via the Settings... window.

Parameters

Horizontal position:

Delay 1 in milliseconds, by default 0 to 1000.

Vertical position:

Delay 2 in milliseconds, by default 0 to 1000.

These values can be changed freely depending on the amount of memory available in your Macintosh (taking into account other open applications). In the case of insufficient memory, an alert panel may appear. You should click on "OK" and either change the delay to a lower value or allocate more memory to the Hyperprism application (Quit Hyperprism, do a Get Info... and increase its memory allocation size).

Diagram of the various delay positions in Multi Delay

Example

Create a new Hyperprism document for the file Gunshot and select a Multi Delay process. Play it and you will hear a single shot fired by a hunting rifle.

In Loop mode, click and move the mouse slowly around in the Blue Window and you'll hear a Western-style shoot-out.

Limitations

Delay and echo effects are sensitive to their input signals. If your input sound is full scale amplitude (i.e., normalized), choose Settings... in the Processes menu and adjust the Pre Attenuation to -3 dB or more before invoking a delay effect. Also, depending on the input signal, you may hear a clicking sound as you move the mouse in the Blue Window. To eliminate this artifact of rapidly-changing delay time, set the Parameter Change Speed (also in Settings...) to 3 or higher.

Clicking sounds can also be heard if:

1. the disk is too slow,

2. the sound file is too fragmented or

3. the speed of the computer is not fast enough to handle the processing algorithms. The three delay algorithms are created by the processor of the Macintosh. It is thus essential to have a fast Macintosh for best results.

See other possible remedies in Appendix C.

The Multi Delay effect performs well on faster Macintoshes but may have difficulty on slower machines. You can speed up the processing by leaving one or more of the attenuation parameters in the Settings... window at 0 (thus saving calculation time that would have been spent rescaling the amplitude of the delayed signal).

Multi Delay is not available in Thru mode.

Echo

The echo effect repeats the input sound many times at the specified delay interval and attenuation. Each echo is attenuated according to the vertical position of the mouse. If the attenuation is 0, for example, the echoes will repeat indefinitely.

A feature of this algorithm is the ability to specify a negative attenuation, typically in the range of -10 to -30 (also applicable to Single Delay). A negative attenuation factor causes the echoes to build in intensity, rather than die out. For effective use, you must control these buildups by adjusting the mouse downward as the echoes begin to reach a crescendo. Using a combination of varying time delays and positive and negative attenuation movements you can create effects similar to recirculating tape delay feedback, such as used in the classic electronic music studios of the 1960s.

Parameters

↔ Horizontal position:

Delay in milliseconds, by default from 0 to 1000. This value can be changed freely depending on the amount of memory available in your Macintosh (taking into account other open applications). In the case of insufficient memory, an alert panel may appear. You should click on "OK" and either change the delay to a lower value or allocate more memory to the Hyperprism application (Quit Hyperprism, do a Get Info... and increase its memory allocation size).

Vertical position:

Attenuation of the repeated signal as a percentage of the total, by default from 0 to 100%. The highest position in the Blue Window represents no attenuation.

Limitation

Delay and echo effects are sensitive to their input signals. If your input sound is high amplitude (or normalized), choose Settings... in the Processes menu and adjust the Pre attenuation value to -3 dB or more before invoking a delay effect. Also, depending on the input signal, you may hear a clicking sound as you move the mouse in the Blue Window. To reduce this artifact, set the Parameter Change Speed (also in Settings...) to 3 or higher. Even with the settings properly adjusted, repeating echoes may introduce transients into smooth sounds.

Clicking sounds can also be heard if:

1. the disk is too slow,

2. the sound file is too fragmented or

3. the speed of the computer is not fast enough to handle the processing algorithms. The three delay algorithms are created by the processor of the Macintosh. It is thus essential to have a fast Macintosh for best results.

See other possible remedies in Appendix C.

Echo is not available in Thru mode.

Ring Modulator

The Ring Modulator considers the input sound to be a *carrier* signal. Using a second signal called a *modulator frequency*, it generates two frequencies that are the sum and the difference of the carrier and the modulator frequencies, each at half the amplitude value of the carrier. Thus, for an input frequency at 400 Hz and a modulator frequency at 25 Hz, the output will be a mixture at 375 Hz and 425 Hz.

Parameters

Horizontal position:

Modulating frequency in Hz, by default 0 to 5000 Hz.

Vertical position:

The presence of the ring modulated sound as a percentage of the total, from 0 to 100%, where 100% represents a completely processed output signal.

Example

Create a new Hyperprism document for the file Bell, play it and then select the Ring Modulator process.

Click at different mouse locations in the Blue Window and hear the changes in the sound.

The Ring Modulator, Frequency Shifter, and Ring Shifter create a wide range of interesting bell sounds, each with their own character.

Frequency Shifter

Like the Ring Modulator, the Frequency Shifter considers the input to be a *carrier* signal. It changes the frequency of each sinusoidal component in the input by a fixed amount according to the mouse position on the horizontal axis. Thus for an input sound having the frequencies 100, 200, 300, 400 and 500 Hz and a modulating frequency of 25 Hz, the output signal has the frequencies 125, 225, 325, 425 and 525 Hz.

Parameters

Horizontal position:

Modulating frequency in Hz, by default from -5000 to 5000 Hz. A negative frequency means that the signal is shifted downward.

Vertical position:

Not used.

Example

Create a new Hyperprism document for the file Bell, play it and then select a Frequency Shifter Process.

Click at different mouse locations in the Blue Window and hear the modifications obtained in comparison with the original timbre.

The Ring Modulator, Frequency Shifter, and Ring Shifter create a wide range of interesting bell sounds, each with their own character.

Ring Shifter

The Ring Shifter combines the Ring Modulator and the Frequency Shifter into one. Ring modulation is applied to a frequency shifted version of the input signal. When the mouse moves, this produces a different type of effect than the plain ring modulator. In particular, you'll hear portamento effects as the tracer changes the modulator frequency.

Parameters

Horizontal position:

The modulator frequency in Hz, by default from -300 to +300 Hz.

Vertical position:

Not used.

Example

Create a new Hyperprism document for the file Bell, play it and then select a Ring Modulator Process.

Click at different mouse locations in the Blue Window and hear the modifications obtained in comparison with the original timbre.

The Ring Modulator, Frequency Shifter, and Ring Shifter create a wide range of interesting bell sounds, each with their own special character.

Pitch Follower

The Pitch Follower tracks the pitch of the input signal. It starts from a previously-analyzed pitch file (created with the Analyze Pitch... operation under the File menu) and applies this pitch envelope to a Hyperprism input file. See the description of pitch analysis in Appendix B.

Parameters

Horizontal position:

Scan speed. A high value causes Hyperprism to read through the pitch envelope rapidly, looping it if necessary to cover the duration of the input file being changed. A value of 0 reads through the pitch envelope at the same speed at which it was originally recorded.

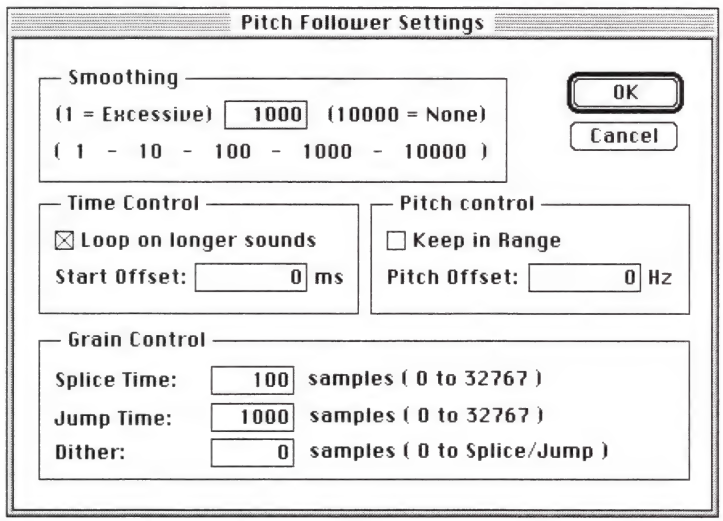
Looping can be switched off in the Settings... dialog. Then if the pitch envelope is shorter than the input file, the pitch of the input file stays at the last value of the pitch envelope until the sound is done playing.

Vertical position:

Ranges from 0 to 30% by default. It corresponds to the degree of pitch shifting. A low value imposes only a slight change in the pitch of the input. A high value imposes a more dramatic change.

Obtaining satisfactory results from the Pitch Follower depends on several factors. First, the parameters of the pitch analysis must be carefully adjusted according to the input sound (see Appendix B). Second, the settings used in the Pitch Follower should be tuned according to the input

sound and desired effect. See the section on Settings at the end of this chapter.


Pitch Follower Settings
Smoothing
(1 = Excessive) 1000 (10000 = None)
(1 - 10 - 100 - 1000 - 10000)
OK
Cancel
Time Control
Loop on longer sounds
Start Offset: 0 ms
Pitch control
Keep in Range
Pitch Offset: 0 Hz
Grain Control
Splice Time: 100 samples (0 to 32767)
Jump Time: 1000 samples (0 to 32767)
Dither: 0 samples (0 to Splice/Jump)

Settings... window for the Pitch Follower effect

Example

As explained at the beginning of this section, in order to use the Pitch Follower it is necessary to previously create a file that contains the pitch variation information. We will then apply this to the file Airport + 1000. In order to do this,

Choose Analyze Pitch... in the File menu and select the sound file named C to G. It contains the notes C, D, F, G.

Therefore, the first and lowest note is C4. The last and highest is G4. Looking at the table in Appendix B notice that these two notes correspond to 262 Hz and 392 Hz, respectively.

To allow room for estimation error, enter 255 as the Min value in the Frequency Range box, and 400 as the Max value in the Frequency Range box. The number of points per second in the Analysis box is correct since the melody is played rather slowly. Click on OK.

A thermometer indicates that the analysis is in progress. By default the analysis file is named C to G-pitch. Click OK unless you want to change the name.

Now create a new Hyperprism document for the file Airport + 1000 and play it.

Select Pitch Follower process in the Processes menu.

At this point Hyperprism asks for a pitch analysis file to apply to Airport + 1000. You should choose the C to G-pitch. file, and click OK.

Now, click in the Blue Window between 0,15 and 0,20.

You should be hearing the announcer as well as the 1000 Hz tone follow the pitch pattern of the sound file C to G.

Note that since the file C to G is shorter than Airport + 1000 you can match their durations by clicking toward the left of the Blue Window where the scan speed is slower.

☞ You can reinforce the pitch follower effect by combining it with an envelope follower effect using multipass processing as follows:

➠ Select.Process to New File... in the File menu in order to create a new pitch-processed Airport + 1000 Hz file which will be given the default name "Airport + 1000 Hz-FX-Untitled P."

➠ Select New in the File menu to create a new Hyperprism document for the new pitch-processed Airport + 1000 Hz file and play it to make sure you are using the pitch-processed rather than the original file.

➠ Read the next section on the Envelope Follower process along with Appendix B on envelope analysis.

➠ Follow the steps given in the Envelope Follower example on page 93, substituting the files "Airport + 1000 Hz-FX-Untitled P-FX" instead of "Provence Ambience" and "C to G" instead of "Train."

➘ You will hear Airport + 1000 played with the pitch and envelope of C to G.

Limitations

Before being accomplished successfully, this effect needs much experimentation according to the type of sound analyzed.

Pitch Follower is not available in Thru mode.

Envelope Follower

The Envelope Follower applies the envelope of one signal (previously analyzed with the Analyze Envelope... item in the File menu) to another signal (selected via New or Open in the File menu). See Appendix B for a description of envelope analysis.

Parameters

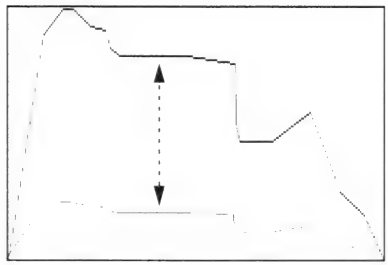
↔ Horizontal position:

Scan speed. A high value causes Hyperprism to read through the amplitude envelope rapidly, looping it if necessary to cover the duration of the input file being changed. A value of 0 reads through the envelope at the same speed at which it was originally recorded.

Looping can be switched off in the Settings... dialog. Then if the amplitude envelope is shorter than the input file, the amplitude of the input file stays at the last value of the envelope until the sound is done playing.

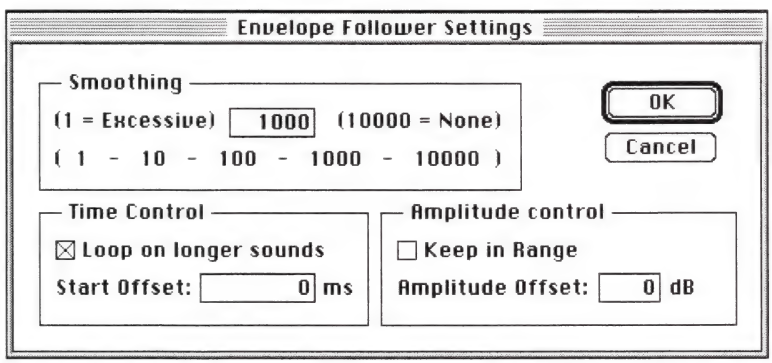
↕ Vertical position:

Range or degree of amplitude change imposed on the input signal, from 0 to 100%, where 100% represents the maximum effect.



Expansion or compression of the amplitude envelope depending on the Range parameter

Obtaining satisfactory results from the Envelope Follower depends on several factors. First, the parameters of the envelope analysis must be carefully adjusted according to sound being analyzed (see Appendix B). Second, the Settings must also be tuned according to the sound one wants to transform. See the section on Settings at the end of this chapter.



Envelope Follower settings

Example

Create a new Hyperprism document for the file Provence Ambience and play it.

Select Analyze Envelope... in the File menu. Choose Train. To accept the default parameters, click OK.

A thermometer indicates that the analysis is in progress. By default the analysis file is named Train-envelope. Click Save or change the name and then click Save.

Now, select the Envelope Follower process and choose the file Train-envelope.

Click in the Blue Window at about 0,80.

You should hear Provence Ambience played with the rhythmic pattern of the sound file Train.

Note that the higher the position of the mouse in the Blue Window, the more the effect is audible as you increase the dynamic range.

IMPORTANT
You can also use the Faders to amplify the signal. To avoid sending too loud a signal to your amplifier, be very careful to remember to put the Faders back to the middle neutral position before playing another sound or switching the listening mode to the original sound.

Limitation
The Envelope Follower is not available in Thru mode.

Doppler

Doppler shift is perceived in natural surroundings when a moving sound passes by a listener at a stationary position. As it approaches and passes, the pitch of the moving sound goes from high to low. A good example of Doppler shift effect occurs when you stand near railroad tracks as a train passes by. Another example occurs when you are sitting in a train and you pass a ringing bell; its frequency increases as you approach it and decreases as you move away. Hyperprism combines this effect with a pan from one channel to another.

☞ This effect transforms a monaural input signal into a stereo output signal. By selecting Process to New File... (in the File menu) one creates a stereo sound file as the output.

Parameters

↔ Horizontal position:

The virtual distance in meters from left to right, by default -1000 to +1000.

↕ Vertical position:

The virtual frontal distance from the source as it passes by, measured in meters from 0 to 300.

Limitations

An effective simulation of Doppler shift requires a very regular mouse movement.

Pitch Time Changer

The Pitch Time Changer lets one vary two aspects simultaneously and independently: the playback speed (duration) and the pitch of the input signal. For pitch-shifting, the algorithm works by loading memory with an incoming signal sampled at rate A and reading out the samples at rate B. The ratio A/B determines the pitch change. To maintain a continuous output signal, samples must be repeated (for upward pitch shifts) or skipped (for downward pitch shifts). Because the output address pointer repeatedly overtakes the input address pointer (for pitch increases) or is overtaken by the recirculating input address pointer (for pitch decreases), the output address must occasionally jump to a new point in the memory. To stretch the time base of the input signal, the algorithm repeats small "grains" (segments) of the input signal, while for time shrinking it deletes intermediate grains.

The sound quality of the Pitch Time Changer is based on the nature of the input signal and on the ratio of pitch change it is asked to perform. Small pitch and time changes tend to generate less audible side effects.

Parameters

↔ Horizontal position:

Speed variation, by default from 0 to 200%, where 0% corresponds to a kind of freezing of the input sound (indefinite duration) and 200 corresponds to a double-speed rendition of the input signal.

Vertical position:

Pitch variation, by default from 0 to 200%, where 0% represents a transposition towards extreme low frequencies and 200 transposes the input sound up an octave.

The Grain Control settings available for the Pitch Time Changer can make a great difference in the effect. See the section on Settings at the end of this chapter.

Limitations

The 200% transposition corresponds to the maximum processing effect possible. Values greater than 200 have no additional effect. For additional pitch and time changing, use multipass processing.

Tremolo

Tremolo varies the amplitude of the input sound smoothly according to a sinusoidal function. For many sustained sounds, a tremolo between 4 and 7 cycles per second adds expressivity, particularly when applied in an increasing manner over the course of the sound. (See the figure on next page.)

Parameters

↔ Horizontal position:

Number of amplitude variations per second, from 0 to 20 by default.

↕ Vertical position:

Degree of amplitude variation as a percentage of the total output, by default from 0 to 100%.

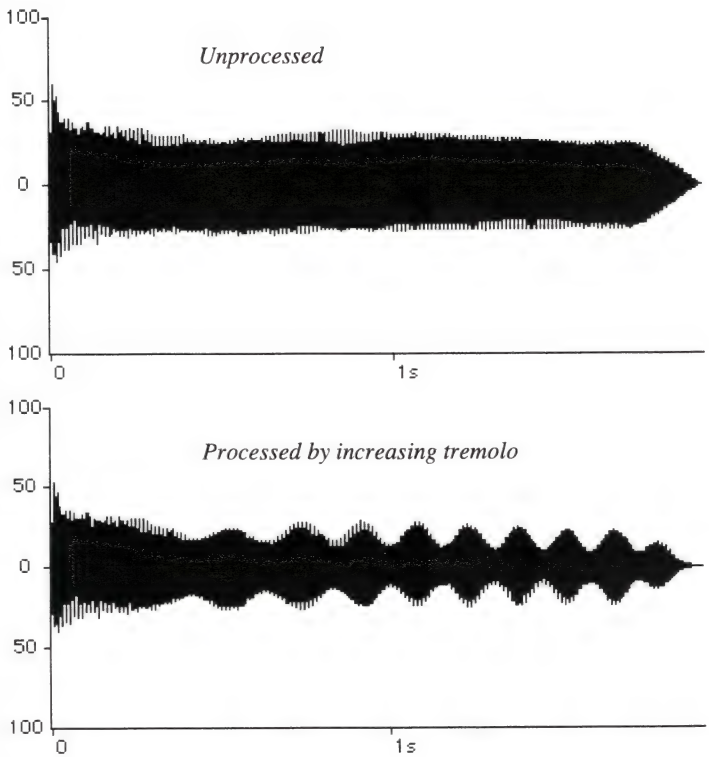
Example

Create a new Hyperprism document for the file Train, play it and select a Tremolo process.

In Loop mode, click in the Blue Window at the bottom left corner and drag the mouse toward the upper left corner, then staying at the top of the window, move slowly to the right.

Listen as your modern Train changes into an old-time steam engine!

Tremolo works very well with Vibrato using multipass processing, particularly on melodic instrumental sounds.



Effect of an increasing tremolo effect on a guitar pluck. The tracer went from 0 Hz to 7 Hz on the horizontal scale, and 0 to 55% on the vertical scale.

Vibrato

Vibrato imposes a time-varying pitch change or modulation on the input signal. The modulation varies sinusoidally at a rate controlled by the user. For many sustained sounds, a vibrato between 4 and 7 cycles per second adds expressivity, particularly when applied in an increasing manner over the course of the sound.

Parameters

Horizontal position:

The number of frequency variations per second, by default from 0 to 20.

Vertical position:

Degree of pitch variation, from 0 to 100% by default.

Example

Create a new Hyperprism document for the file Bell, play it and select a Vibrato Process.

Click in the Blue Window at about 4,7.

You can hear the effect of a normal vibrato.

Try using this effect in conjunction with Tremolo using multipass processing.

Stereo Dynamics

Stereo Dynamics spatializes the input sound, positioning it in a virtual space situated in front of the listener. It moves the input sound from front to back and from left to right, depending on the position of the tracer in the Blue Window.

Parameters

↔ Horizontal position:

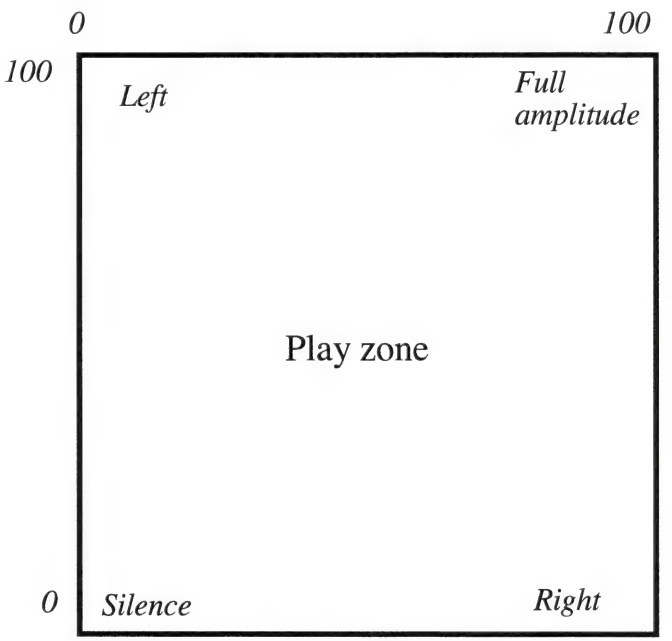
Right - left gain, which controls the proportion of one channel relative to the other. For example, moving the tracer to the right will increase the proportion of the right channel in the output.

↕ Vertical position:

Right + left gain, that is, the overall amplitude of both channels.

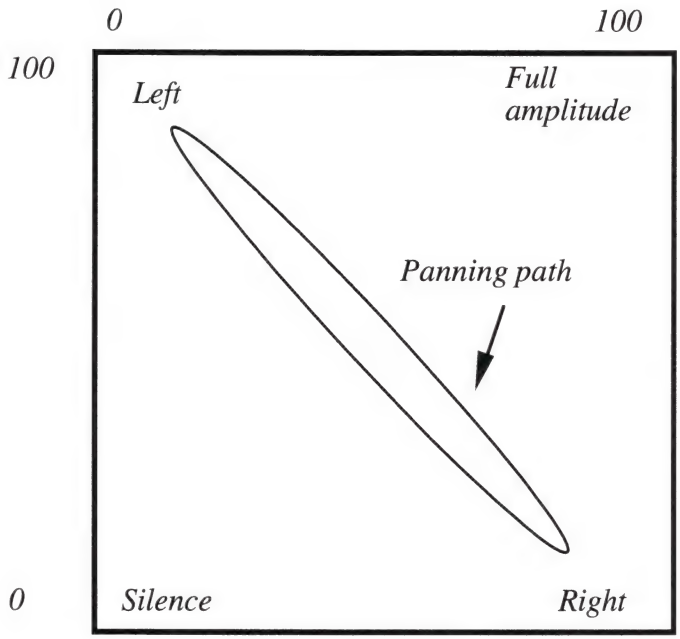
☞ This effect transforms a monaural input signal into a stereo output signal. By selecting Process to New File... (in the File menu) one creates a stereo sound file as the output.

The figure below shows the layout of the play zone for the Stereo Dynamics effect.



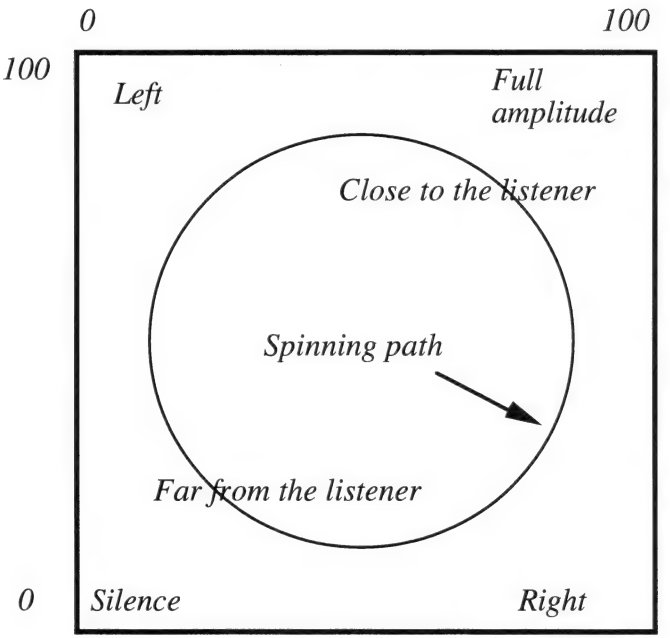
Default layout of the play zone for Stereo Dynamics

Notice in the next figure that as you move the tracer from top left to bottom right, you pan the sound.



Tracer path for a panning motion

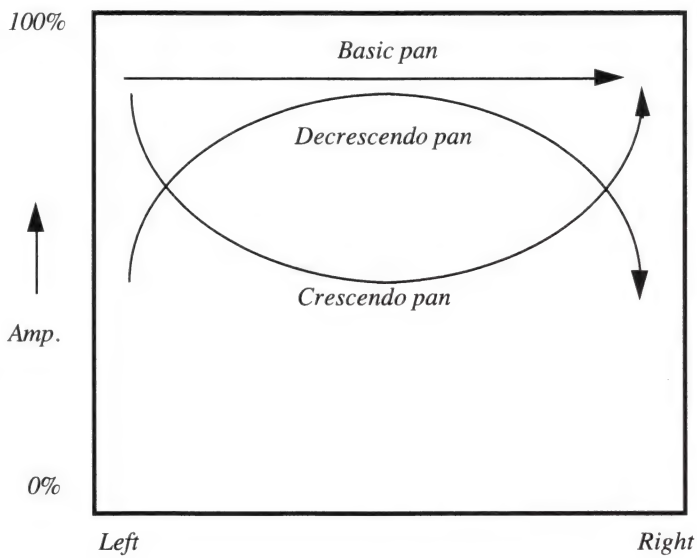
As you move it from top right to bottom left, you attenuate it. So if you combine the two motions, as the following figure shows, the sound appears to spin in a circle in front of the listener situated precisely between two loudspeakers. This is also an excellent effect on headphones.



Tracer path for a spinning motion

Balance

Balance pans the input sound from left to right, following the mouse position, and combines this with a time-varying attenuation, if desired. For a more dramatic panning motion, start with the signal attenuated in one of the channels; as you pan, put in a crescendo as the sound passes through the center. The opposite effect, a decrescendo pan, may also be effective, since it creates a "hole in the middle" effect.



Crescendo and Decrescendo pans

☞ This effect transforms a monaural input signal into a stereo output signal. By selecting Process to New File... (in the File menu) one creates a stereo sound file as the output.

Parameters

↔ Horizontal position:

The position of the sound in the plane from left to right, by default from -100 to +100% where 0 corresponds to the center position.

↕ Vertical position:

Attenuation of the sound from 0 to 100%, where 100% means no attenuation.

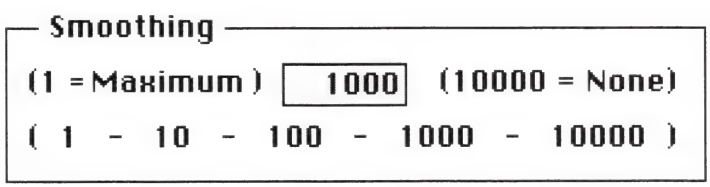
Settings

For each effect possible in Hyperprism, you can make certain adjustments beyond those in the Blue Window. This is a very important feature for controlling Hyperprism effects precisely. When a window is active, select Settings... in the Processes menu. A dialog window appears, letting you adjust the various parameters of a particular effect algorithm.

Smoothing

Smoothing is available for the following effects:

Low Pass	Phaser	Doppler
High Pass	Flanger	Pitch Time Changer
Band Pass	Chorus	Tremolo
Band Reject	Ring Modulator	Vibrato
	Frequency Shifter	Stereo Dynamics
	Ring Shifter	Balance

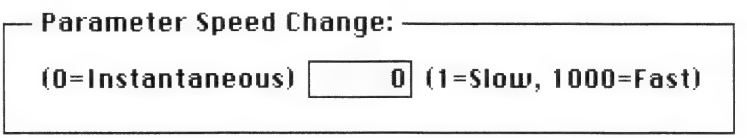


Smoothing affects the values that Hyperprism reads in from the tracer on the Blue Window. When the smoothing factor is set to low, Hyperprism averages values read from the mouse position.

Thus, smoothing makes an effect glide from one parameter setting to another, rather than jumping abruptly. In musical terms, one could say that it functions as a kind of portamento between parameter settings.

Smoothing can be modified between 1 (maximum smoothing) and 10000 (no smoothing). The scaling is logarithmic, as indicated by the proposed values in the Settings... window. So each power of ten "halves" the effect of the smoothing. Thus, a value of 100 is half the smoothing of 10.

Parameter Speed Change

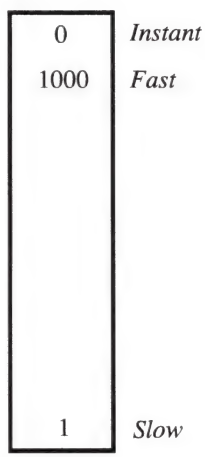


Parameter Speed Change is available for the following effects:

Single Delay
Multi Delay
Echo

☞ When you move the mouse using these effects, this movement corresponds to varying the reflection time slightly, as if the wall reflecting the source is moving. This movement is not natural, and can introduce clicks that affect the audio quality of the processing. The role of Parameter Speed Change is to give a virtual inertia to mouse movements in order to lessen the unwanted side-effects. The greater the inertia, the slower the change from one delay setting to another, avoiding the glitches associated with the "fast moving wall" effect.

A value of 0 is "magic" in the sense that it makes all delay values coming from the Blue Window take effect instantaneously. Between values of 1 and 1000 there is a scale ranging from very slow (1) to 1000. The figure below depicts this scale.



Scale of the parameter speed change

☞ Parameter Speed Change can be used as an effect in itself, in the sense that a variation in it progressively changes a slight delay value into a large delay without further intervention of the user. Click on a value of about 50 ms and then click on a value approaching 1000 ms (or more if you have sufficient RAM in your Macintosh). Notice how the delay progresses depending on different settings in the Parameter Speed Change parameter.

The default value is 10. A value of 1000 corresponds to a quasi immediate transition from one value to another. A value of 100 corresponds to a slow, progressive transition from one value to another.

Pre Attenuation

Pre attenuation is available on the following effects:

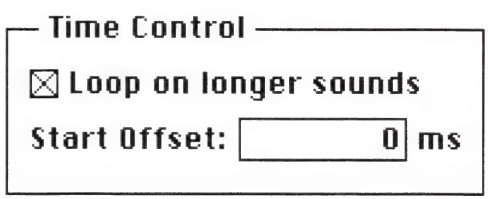
Single Delay
Multi Delay
Echo

Loud signals may cause overload in the output signal when they are delayed and added to themselves. Keep in mind that one is repeatedly compounding the amplitude at each delay time. Pre attenuation lets one attenuate the input signal before it is treated, possibly avoiding overload problems.

The default pre attenuation value is 0 dB. Valid attenuations are between 0 and 100 dB. An attenuation of 1 to 6 dB is often sufficient to avoid overload. For smooth input sounds, a greater attenuation factor may be necessary to avoid the introduction of echo transients.

☞ Note that the editable boxes in which you enter the pre attenuation values already have a minus sign (-) in front of them so that you don't have to type it.

Time Control



Time control settings are available for:

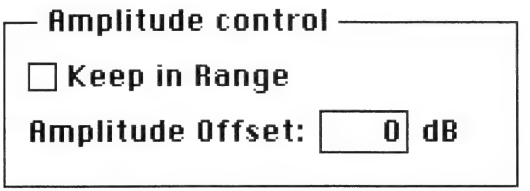
Pitch Follower
Envelope Follower

Loop on longer sounds:
repeats the envelope in case the input sound is longer than the analyzed envelope.

Start offset:
offset factor into the envelope. An offset factor of 1 sec will start reading the envelope at its 1 sec point. This does not imply a delay in the effect, which takes place immediately.

Amplitude Control

Amplitude control settings are available for the Envelope Follower effect.



Keep in range:
overrides the amplitude offset to clip the maximum amplitude to within the original range.

Amplitude offset:
value in dB added to the overall intensity of the amplitude envelope.

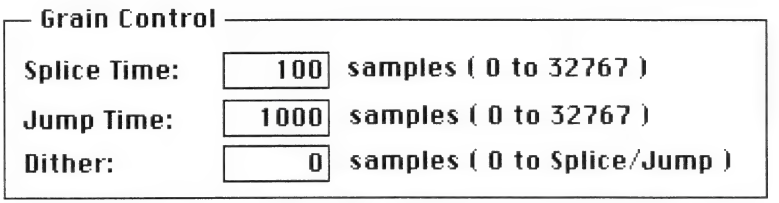
Pitch Control

Pitch control settings are available for the Pitch Follower effect.

Pitch offset:
factor to be added to the analyzed pitch envelope. For example, if the original pitch range was 100 to 400 Hz and the pitch offset is set to 100 Hz, the new pitch range is 200 to 500 Hz.

Keep in range:
Overrides the pitch offset to clip the maximum frequency to within the original range. In the above example, the resultant output range would be 200 to 400 Hz, even with a pitch offset of 100 Hz.

Grain Control



Grain settings are available for:

Pitch Time Changer
Pitch Follower

These low-level controls can make a great difference in the output of the effect.

Jump time:
How often the waveform segmenting algorithm jumps forward or backward; can be considered the "grain size."

Splice time:
Crossfade time (in samples) between grains.

Dither:
Randomizes at the specified factor for jump time. Use with caution in order to avoid a constant modulation "buzzing" effect.

There are no "magic" settings for these parameters. Only by experimenting with a given input sound can you explore the range of sound transformation possibilities these settings can realize.

Grain Control for the Pitch Time Changer

The Grain Control settings can have a strong effect on the nature of the output of the Pitch Time Changer. Depending on how these settings are adjusted, the effect can range from echoing to "granulation" to obliteration of the identity of the input signal. Here are some guidelines.

Speech makes an excellent test of the algorithm, thus:

- Select a speech file for input.
- Set the speed parameter to between 5 and 50%, while keeping the pitch parameter constant at 100%. For testing purposes, do not adjust these controls while manipulating the Grain Control settings.
- Click the mouse in the Blue Window and play through the file once at a fixed tracer position.
- Now select Settings... in the Processes menu, adjust the grain control parameters and click "OK" when you have made the adjustments.
- Click on Play in the playback palette.

As you set the jump time below 1000, notice how the effect becomes more buzzy. A jump time of less than 200 turns into a metallic drone. On the other extreme, jump times over 10000 turn into repeating echo effects.

Notice that the effect of the splice time parameter is linked with the jump time. Generally, for longer jump times, the splice time should be no less than one tenth of the jump time in order to avoid certain noises. In other cases, the splice time can actually be longer than the jump time. This effect is particularly effective for jump time values lower than 1000.

Dither randomizes the jump time. It can reduce the harmonic buzzing effect, but it may also contribute "bumpiness" or objectionable noise. For longer jump times, it randomizes the echo time.

A Hyperprism Installation and Configuration

This appendix describes the procedure for installing and configuring Hyperprism on your Apple Macintosh computer.

Hardware and Software Requirements

Before using Hyperprism, please make sure that the following conditions are satisfied completely:

❶ *You should possess a basic knowledge of the Macintosh software environment. This includes knowing how to use the mouse, windows, menus, etc. If these terms are strange to you, please consult your Macintosh documentation.*

❷ *Your Macintosh must have a floating-point unit (FPU) and support Digidesign sound cards. (However, Hyperprism does not run on the SE30.)*

❸ *The operating system must be System 7.0 or later.*

❹ *The DigiSystem INIT installed in your system (in the Extension folder) must be version 2.3 or higher.*

❺ *Your Macintosh must have at least 5 Mbytes of random-access memory (RAM). If you want to create long delay effects you will need more memory, since Hyperprism uses available Macintosh memory for its delay lines. We recommend that your Macintosh have at least 8 Mbytes of RAM total. You can check the amount of memory allocated to the Hyperprism application by selecting the Hyperprism icon and typing Command-I. The program as distributed works with 2 Mbytes of RAM, which allows delay times beyond 5 seconds for monaural files at a 44.1 KHz sampling rate. Allocating more than 2 Mbytes to the Hyperprism application allows the delay times to be extended much further.*

❻ *Hyperprism runs only on Macintoshes with Digidesign cards. Any Digidesign card works, from the oldest Sound Accelerator to the latest Pro Tools system, including Audiomedia I and II.*

❼ *Finally, verify the analog and digital connections in your configuration.*

Before Beginning

When opening your Hyperprism package you will find the following ingredients:

- ✔ One registration card
- ✔ One user's manual
- ✔ One license agreement on the envelope containing:
  - ✔ One Master Key Disk with utility programs
  - ✔ One Hyperprism disk with the Hyperprism application
  - ✔ Three diskettes with sound file examples

IMPORTANT

Please fill out your registration card and mail or fax it to Arboretum Systems as soon as possible in order to:

- facilitate your getting technical support
- be informed of new developments and upgrades
- help shape new developments and upgrades

Note that in order to receive telephone support and technical assistance from Arboretum Systems, you must be a registered user.

☞ IMPORTANT

The Master Key Disk is very precious. It contains the program to authorize Hyperprism on a hard disk and also the copy protection codes. Therefore,

You Should Never Erase or Reformat Your Master Key Disk.

If you do, you will void Arboretum's commitment to the terms of the License agreement.

This Master Key Disk allows you to:

- authorize (two times) or deauthorize (two times) Hyperprism on a hard disk, or

- launch Hyperprism from your hard disk without authorizing it by inserting the Master Key Disk when requested.

☞ IMPORTANT:
Make sure that the first thing you do with your software is to make a backup copy of your Master Key Disk and of your Hyperprism Disk (including all the files these disks contain) by icons on two 800 Kb diskettes and keep these diskettes in a safe place. Note that the backup of your Master Key Disk will not give you extra authorization credits: it will only allow you to replace the files on your original diskettes in the event those files are accidentally erased or become corrupted.

Copying the Example sound files

The Examples Sounds diskettes included in your package are provided to help you familiarize yourself with Hyperprism.

Copy them by simply inserting the diskettes and dragging all their contents onto your hard disk. If you want, you can create a folder named "Example sound files" on your hard disk.

Keep a copy of these files on your hard disk until you are familiar with the software. The Guided Tour of Hyperprism in Chapter 3 uses several of the example files, as does the Description of Hyperprism Processes in Chapter 4.

Here is the list of these file examples:

File Name	Disk #
Airport + 1000	1/3
Gunshot	1/3
Beethoven	1/3
Provence Ambience	2/3
Train	2/3
C to G	3/3
Bell	3/3

Make sure that you have enough space to copy these file onto your hard disk. They require a total amount of about 3 Mbytes.

Installing Hyperprism

IMPORTANT
Please do not launch Hyperprism from its diskette. Copy it first onto your hard disk and launch it from there.

Copy your Hyperprism application onto your hard disk.

To do so, simply drag the Hyperprism application from the Hyperprism Disk to your hard disk, as you would transfer any other file.

IMPORTANT:
Make a backup copy of your Master Key Disk and of your Hyperprism Disk (including all the files these disks contain) by icons on two 800 Kb diskettes and keep these diskettes in a safe place. Note that the backup of your Master Key Disk will not give you extra authorization credits: they will only allow you to replace the files on your original diskettes in the event those files are accidentally erased or become corrupted. After an authorization, keep your Master Key Disk write-protected to avoid erasing it accidentally.

To use Hyperprism, you have your choice of either authorizing or not authorizing Hyperprism to launch directly from the hard disk of your choice.

Using Hyperprism Without Authorizing it on your Hard Disk

You may choose to use Hyperprism without authorizing it on your hard disk, thus keeping your credit of two authorizations intact.

To launch Hyperprism without authorizing it on your hard disk, double-click its icon on your hard disk as you would for any other application.

☞ In this case, the first time you launch Hyperprism a window will appear asking you to insert your Master Key Disk and, after a brief instant, your key disk will be ejected and Hyperprism will launch.

☞ In addition, the very first time you launch Hyperprism you'll be requested to enter your Name, Company (optional) and your serial number. You'll find this number on the label of your Master Key Disk.

☞ IMPORTANT
Unless you authorize Hyperprism on your hard disk, your Master Key Disk will be requested each time you launch Hyperprism after having started or restarted your Macintosh.

☞ If Hyperprism does not start, or if anything else seems wrong, consult Appendix C at the end of this manual.

Authorizing Hyperprism on your Hard Disk

If you want to launch Hyperprism from your hard disk without having to insert your Master Key Disk every time your Mac has been restarted, then use the procedure below:

➠ Insert your Master Key Disk, double-click on the application called **Authorizer** and follow the instructions given by this program.

Window of the Authorizer program at first launch

❶ Select the hard disk onto which you want to authorize Hyperprism, using the **Target Volume** pop up menu (in our example, the hard disk mentioned is named "HARD DISK").

❷ Click on the **Authorize** button.

❸ Wait for the success message in the **Message** box.

❹ Note that the number of **Remaining** authorized installations has been decreased by one.

❺ **Quit** the program Authorizer.

You are now done with this procedure and may launch Hyperprism at will.

☞ The very first time you launch Hyperprism, you'll be requested to enter your Name, Company (optional) and your serial number which is printed on the label of your Master Key Disk.

☞ You may use disk optimizers without risking your Authorization credits.

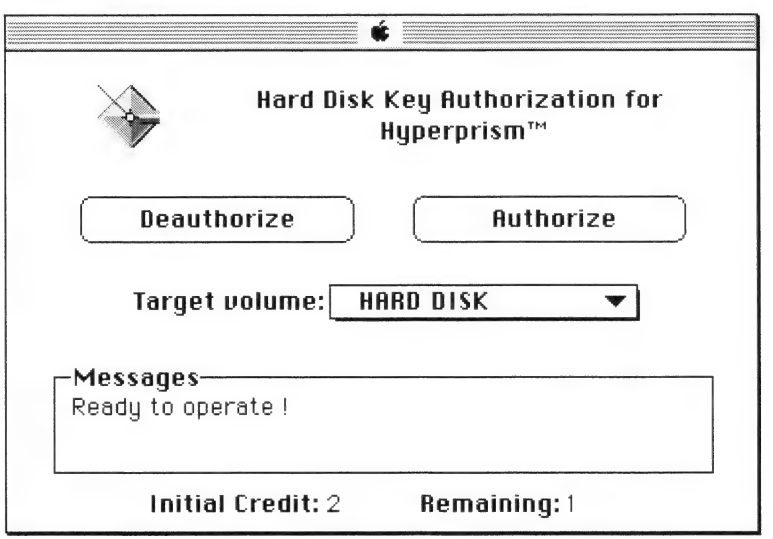
☞ If Hyperprism does not start, or if anything else seems wrong, consult the Appendix C at the end of this manual.

Deauthorizing Hyperprism From your Hard Disk

If for any reason you need to reformat your hard disk, or if you wish to authorize Hyperprism on another hard disk, you will need to deauthorize Hyperprism first in order not to loose one of your two authorization credits.

The procedure is as simple as the authorizing procedure:

➠ Insert your Master Key Disk, double-click on the application called **Authorizer** and follow the instructions given by this program.



Window of the Authorizer program after one authorization

❶ Select the hard disk from which you want to de-authorize Hyperprism using the **Target Volume** pop up menu (in our example, the hard disk mentioned is named "HARD DISK").

❷ Click on the **Deauthorize** button.

❸ Wait for the success message in the **Message** box.

❹ Note that the number of **Remaining** authorized installations has been increased by one.

❺ **Quit** the program Authorizer.

To re-authorize Hyperprism on your hard disk, follow the procedure explained earlier in this section.

Configuring Your Hyperprism System

If you have more than one sound card, you'll need to configure Hyperprism. This is a two-stage process: you first tell Hyperprism which card you would like it to use and then you specify the parameters of your choice as you would with your Digidesign software.

(If you only have one card, this procedure is not necessary.)

Step 1 **Select Device**

Choose Select Device... in the Setup menu.

A list of one or more cards installed in your Macintosh appears.

Select a card and click on OK.

If you have more than one sound card, you may need to make this selection each time you start Hyperprism. (If it selects the card you want by default then it won't be necessary.)

Step 2 **Setup Device**

Choose Setup Device... in the Setup menu.

A Setup Device... window appears, allowing you to set the selected card's specific parameters.

Now your system is configured.

Using File Mode

Hyperprism works in two modes: File mode and Thru mode. File mode treats sound files on disk, while Thru mode processes sounds fed "live" directly into and out of your sound card. Both modes are explained in detail in Chapter 3.

To work in File mode, you need to already have sound files on disk. You can record sound files with Sound Designer II or obtain them by other means (synthesizing them with Digidesign's TurboSynth program, importing them from a sampler through the Passport Designs' Alchemy sound editor, from Studer Dyaxis I audio workstation files converted to AIFF by Alchemy, etc.).

The acceptable input file formats to Hyperprism are Sound Designer II (SD II) and Audio Interchange File Format (AIFF). The acceptable sampling rates are 44.1 and 48 KHz. If you have sound files that have been created at other sampling rates, convert them to these sampling rates before trying to process them with Hyperprism. Sound Designer II's SR Convert command (under its DSP menu) can perform this operation.

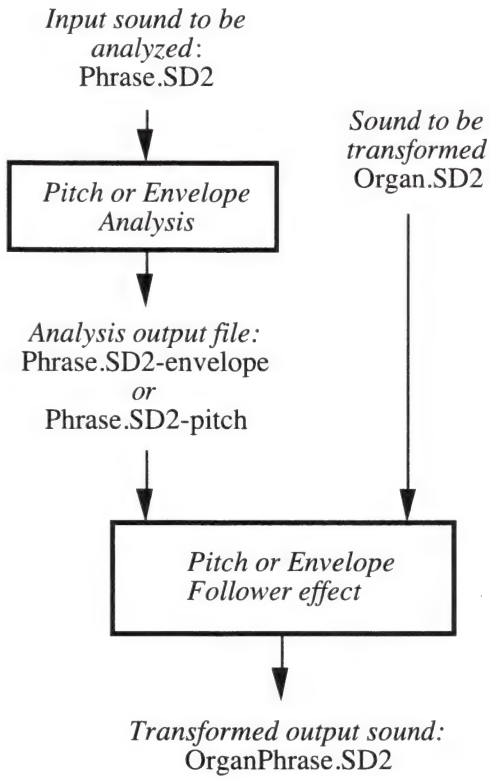
B

Envelope and Pitch Analysis

Amplitude envelope and pitch analyses are necessary procedures before one can use the pitch and envelope followers. To invoke pitch or amplitude envelope analysis, select Analyze Pitch... or Analyze Envelope... in the File menu. This opens a window showing various analysis parameters. With some experience, you should be able to adjust these parameters to achieve optimum results. The output of pitch and envelope analyses are stored in separate Hyperprism documents appended with the extension "-envelope" or "-pitch."

Icon of document created after an envelope analysis (left). Icon of document created after a pitch analysis (right).

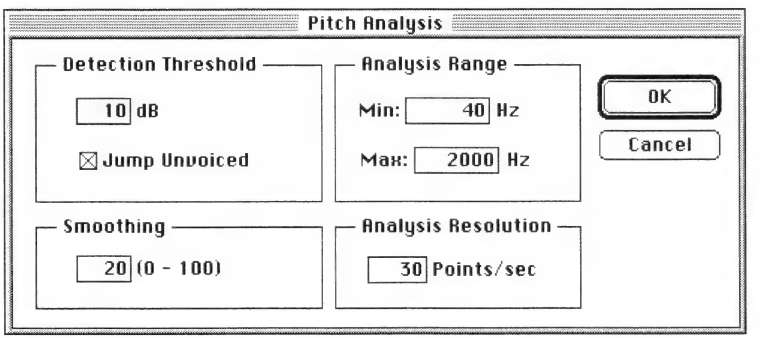
The following figure depicts the process of analysis and transformation:



Analysis and transformation by envelope and pitch analysis in Hyperprism

Best results can be obtained by analyzing monaural files, since Hyperprism combines the left and right analyses of stereo files into one analysis output file. The Pitch and Envelope Followers read only a single analysis file at a time. You can tell them to read a combined two-channel analysis or a one-channel analysis. This single envelope will be applied to both channels, in the case of a stereo input file.

Pitch Analysis Parameters



Pitch analysis parameters

Pitch analysis has several parameters:

Detection threshold:
Lowest amplitude range, below which the pitch follower assumes there is no signal to analyze, so no analysis is attempted.

Jump Unvoiced:
Skips noisy parts of the input signal. Important in analyzing speech and the singing voice. When Jump Unvoiced is selected, the pitch analysis skips over noisy parts of the input signal corresponding to consonants and focuses exclusively on sustained vowel-like sounds.

Analysis range:
Frequency range of the analysis. This parameter must be set carefully to the range of the input signal. Best results are obtained when the specified range is slightly larger than the actual range. If the analysis range is too wide, it is likely that the pitch analyzer will make octave errors. That is, it will recognize the right pitch class, but estimate it an octave or two lower or higher. In this case the "melody" will hardly be recognizable.

Refer to the following tables to help you set the analysis range parameters.

Table of Pitch Classes

C	262 Hz
C#	277
D	294
D#	311
E	330
F	349
F#	370
G	392
G#	415
A	440
A#	466
B	494
C	523

For the human singing voice, typical ranges are the following:

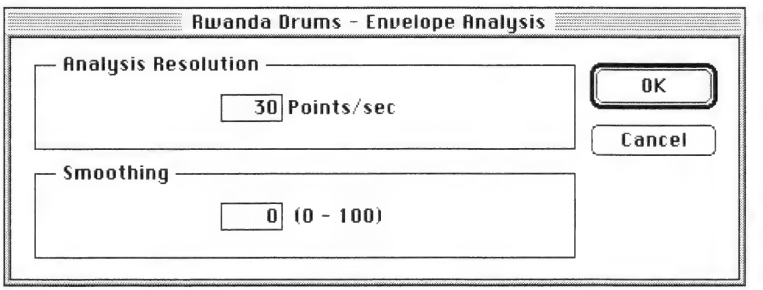
Bass:	82	to	311Hz
Baritone:	110	to	392 Hz
Tenor	140	to	494 Hz
Alto:	196	to	698 Hz
Soprano:	262	to	1046 Hz

Smoothing:
Degree of smoothing between pitches in the output (a value between 0 and 100, where 100 means maximum smoothing). Smoothing can sometimes help eliminate glitches caused by errors in the pitch detection.

Analysis resolution:
Indicates the number of analyses per second, akin to the sampling rate of the melody (not the frequencies in the melody!). By default, the resolution is set at 30, which corresponds to 15 notes per second (since one wants to detect the beginning and ending of each note). But to track a smoothly changing pitch, the analysis resolution should be increased to 50 or more. Increasing the resolution also increases the calculation time, however. Note that by lowering the analysis resolution it is possible to "quantize" the pitch changes, eliminating vibrato and smooth changes.

Once the analysis is done, Hyperprism provides a wide range of parameters in the Pitch Follower effect for remapping the analyzed pitch envelope onto a new sound file. See the description of this effect and the example in chapter 4.

Envelope analysis parameters



Envelope analysis parameters

Amplitude envelope analysis has just two parameters:

Analysis resolution:
Number of points per second, akin to sampling rate of the envelope. If the input sound has a tremolo, this value should be at least twice the frequency of the tremolo in order to accurately sample the tremolo.

Smoothing:
Extent to which sharp changes in the amplitude of the input signal are smoothed out in the analyzed envelope signal.

C

Troubleshooting

This section will help you solve the most common problems which may occur while using Hyperprism. If none of the solutions proposed seem to be effective, then contact your dealer or Arboretum Systems.

If Hyperprism does not work well at all !

A good reason why Hyperprism would not work well or at all would be that your version of the *DigiSystem Init* extension is obsolete. This extension must be version 2.3 or higher. If your version is obsolete, replace it with the one provided on your Utilities disk as follows:

- Drag the *DigiSystem Init* icon from the floppy to your System folder on your hard disk where it will be placed automatically into the Extension folder.
- Click OK to replace the previous extension with the new one.
- Restart your Macintosh and try again.

If there is "No FPU"

Hyperprism can only run on a Macintosh with a *Floating Point Unit* (or arithmetic co-processor). If your Macintosh does not have an FPU, you may have the option of contacting your local Apple dealer to have it installed.

If there is "No Card"

You may get the error message "You need a DSP card to run this program" even when you have a card properly installed if there is a card access conflict between Hyperprism and your Sound Designer II or Pro Tools software.

Here is the best way to solve this problem:

- Quit all applications using a Digidesign card.
- Open your System Folder and locate the file named *DigiSetup*.
- Drag it to the Trash and empty it.
- Create a new DigiSetup file by launching Sound Designer II or Pro Tools.
- Verify from within Sound Designer II or Pro Tools that you have sound coming in and going out of your system.
- Launch Hyperprism.

☞ Hyperprism does not memorize your hardware system configuration. It uses the current setup configuration for input/output and for digital format. Although it does automatically select the sample rate required by the sound file it is about to play and/or to process, if there is a card access conflict, you must redo your setup by launching Sound Designer II or Pro Tools after having trashed the DigiSetup file to create a new one.

If no sound can be heard

If you have no sound when you click Play, check the following:

- In Thru mode, you need to have an input source connected with sound playing and the device window in front.

- In File mode, you need to have at least one Hyperprism document open.

- Your setup may not be valid (see the **If there is no card** section above).

- Check to see if you have turned on the power of your peripheral.

- Your hardware may not be able to play the sample rate of the file (e.g. Sound Tools I without peripheral or only AD IN cannot read a sound recorded at a 48 K sample rate).

☞ You can always tell if the sound is taken into account by Hyperprism when the Peak Meters' LEDs light up.

If you hear clicking noises

This may occur when you are using the delay effects (Single Delay, Multi Delay and Echo) and you move the mouse too quickly in the Blue Window or your Macintosh is not fast enough. See the description of the delay effects in Chapter 4. See also the section **How to optimize performance** in this Appendix.

If no command is available

If you experience a problem with the application in a way that none of the commands (buttons, menus, etc.) seem to be available or respond, you can try to recover with the "Force Quit" command by simultaneously pressing Option and Command and, when those are down, the Esc key. This will open a dialog proposing you to quit. You'll then return to the Finder.

Please try to describe what happened and report the problem to Arboretum Systems.

If your sound is distorted

You may have distortion when switching from Original to Variation on one or both channels while a sound is being processed. This will occur when using the following effects:

Pitch Time Changer
Doppler
Vibrato

With these effects, the only way to hear the original and the processed sound is to click on the Stop button, change the listening mode, and then click on Play again.

If colors go wrong

If you switch from Hyperprism to a graphic application or change the settings... in the Monitors window and then come back to Hyperprism, you may find that the Faders have strange colors. This will not alter their function but will remain until you Quit and launch Hyperprism again.

Extensions problems

Strange behavior may sometimes be caused by System extensions and Control Panels.

There is no known conflict between Hyperprism and any System extensions or Control Panels. If, however, you want to find out for sure, remove all installed extensions except DigiSystem INIT using the init management software of your choice. (You can place the Extensions in a folder temporarily). You can then retry the configuration by installing the extensions one by one.

Note that quite a few of these may have impact on performance. You may want to have as few as possible installed.

If you can't open a sound file with Pro Tools

Pro Tools may not be able to open some stereo files generated by Hyperprism if their titles becomes too long when Pro Tools adds the suffix *left* or *right* to them (not a problem, therefore, with mono files). This is typical when you generate a file with a default title such as *Jazz Guitar-FX-Untitled Balance* for which only one more character is enough to go beyond the maximum number of characters (31) allowed by your Macintosh System.

To remedy this situation, just shorten the names of the files you generate with Hyperprism.

This will be fixed in a future version of Pro Tools.

How to optimize performance

Hyperprism performs best on faster machines.

Also, faster machines and disks can handle digital audio data throughput with less strain. Best performance for echo and delay effects is realized on Macintosh models IIfx and faster.

For optimum performance, use the 44.1 KHz sampling rate, and process monaural files. Make sure your disk is not too fragmented. A fragmented disk makes it more difficult to read a sound file since it is scattered in small fragments. We recommend the use of a disk defragmenter program.

The other ways to improve performance are listed below:

- Do not use Virtual Memory
- Use few System extensions
- Do not use Appletalk
- Use 256 colors or less
- Close the Time Display and/or Peak Meters palettes

Viruses and Authorization Credits

This protection scheme and the installation program were designed to be no more of a problem than a non-copy-protected program in case of virus attacks on your target volume. Of course, you need to have made a backup copy of your Master Key Disk by icons from the Finder before your files become corrupted, so make sure that the first thing you do with your software is to back it up.

If you find a virus on your target volume, just reboot the machine with a disk containing a good System and a good copy of the installation program, launch the installation and insert the key-disk when prompted in order to remove installation from the target volume, even if the volume and the Master Key Disk are infected.

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